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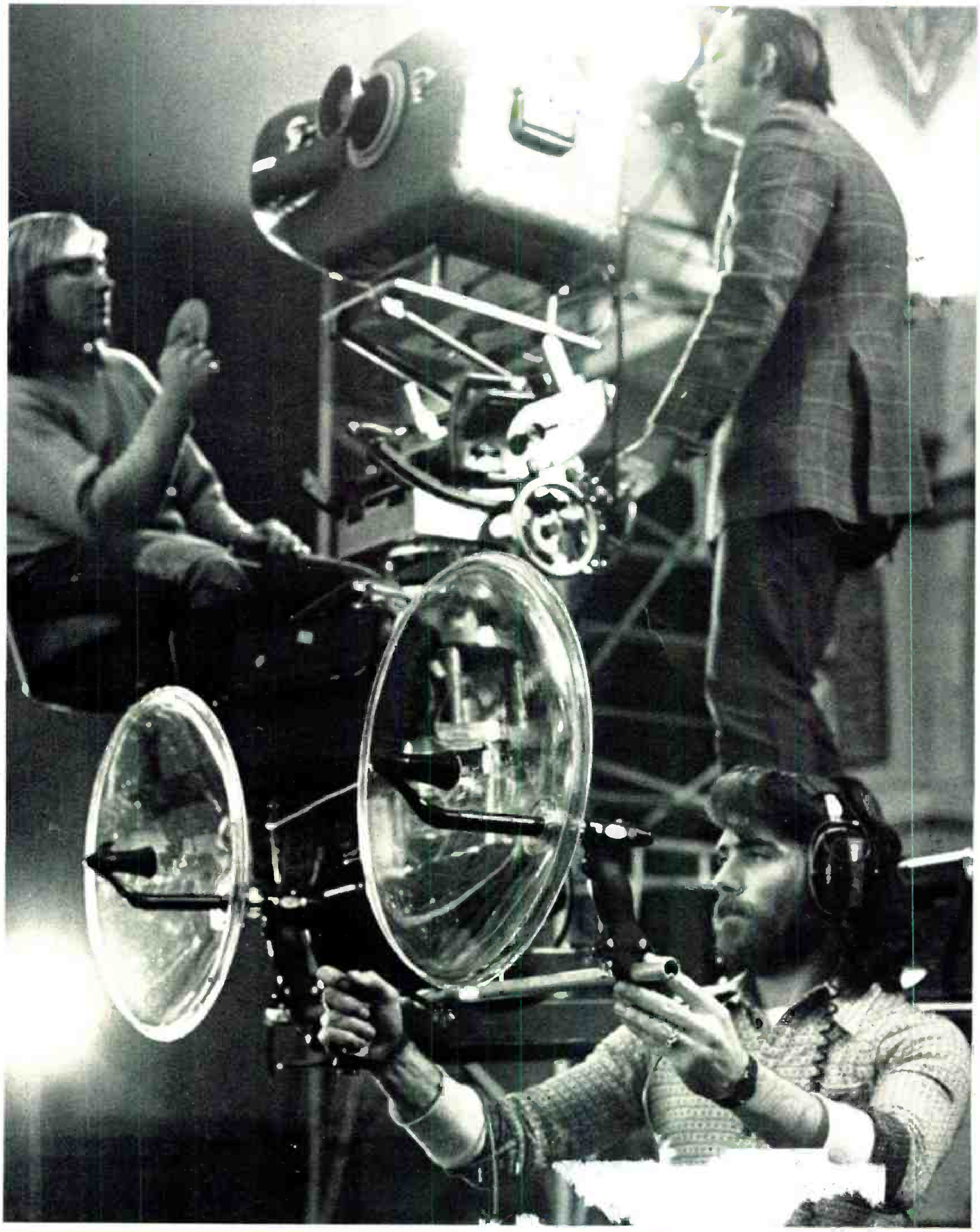
THE SOUND ENGINEERING MAGAZINE

AUGUST 1974 $1.00



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COMING NEXT MONTH

● EQUALIZATION IN MAGNETIC RECORDING describes a method used to evaluate recording non-linearity by means of modulation characteristic as well as data concerning amplitude distribution over the audio spectrum. It was written by a Bulgarian and originally appeared in a Russian publication. The author is M. Boyanova and the article has been translated by George Alexandrovich.

Short but interesting, describes James W. Burlingame's Almost Something-For-Nothing Power Supply. When you need a negative voltage when only a positive 12 V car battery exists, you will be glad to have saved this circuit.

What happens to balances when a live symphony orchestra must be combined with magnetic tape sound and other electronic effects? Stephen H. Lampen has the answer, at least insofar as a performance of the San Francisco Symphony is concerned, in AMPLIFYING AN ORCHESTRA.

And there will be our regular columnists: Norman H. Crowhurst, Martin Dickstein, and John Woram. Coming in **db, The Sound Engineering Magazine.**

ABOUT THE COVER

● The parabolic mic has been around for a long time but it is only now coming into its own as a serious recording tool. Here are two in field use, a function they serve well.

db

THE SOUND ENGINEERING MAGAZINE

AUGUST 1974 VOLUME 8, NUMBER 8

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letters

The Editor:

In two recent advertisements[1,2] in your magazine, statements are made in regard to multi-pattern microphones which may be misleading to some readers. Under the heading, *The Truth About Patterns* is the following: "It is true that all of the switchable characteristic microphones also have an omni-directional pattern position, but it is formed through the electrical combination of two cardioids and therefore largely behaves like a directional microphone."

While the statement may reflect undesirable properties of some multi-pattern microphones, the limitations cited are certainly not true for all multi-pattern microphones. Those microphones in which the omni-directional pattern ". . . largely behaves like a directional microphone" are manufactured after the design disclosed in the Grosskopf patent.[3] In this design, the pattern change is achieved by the electrical switching (phasing) of two cardioids. The system has been described by Bauch[4] and Eargle.[5]

The condenser microphones manufactured under the Shoeps patents[6] do

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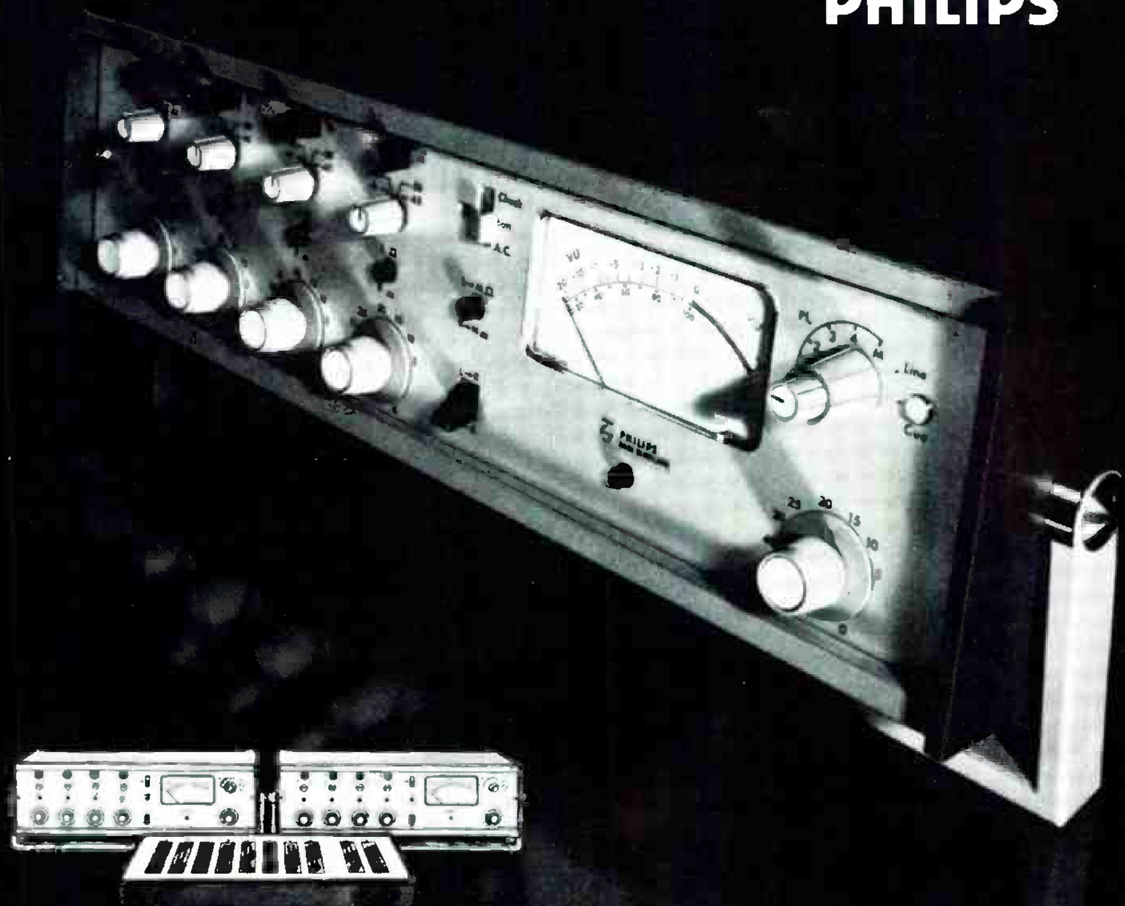

not have these limitations. In the Schoeps microphones, not only does one have the advantage of acoustically pure cardioid and figure-eight patterns, but also the advantage that the omni-directional pattern is a pure pressure transducer and is therefore ". . . completely free of the proximity effects such as popping, low end boost, and high-end edginess . . ." associated with the pressure-gradient transducers. The principal on which the Schoeps microphones change patterns is the mechanical switching of the acoustical chambers behind the diaphragm. The capsule is basically a pure pressure omni which is modified to a cardioid and to a figure eight. Both of the Schoeps patents describe methods by which these acoustical pattern changes can be achieved by electrical switching.

REFERENCES:

1. db Magazine—December, 1973, p. 6.
2. db Magazine—January, 1974, p. 6.
3. H. Grosskopf—U.S. Patent #2678967, assigned to Nordwestdeutscher Rundfunk, Hamburg, Germany
4. Bauch, F. W. O.—"New High-Grade Condenser Microphones," J.A.E.S., Vol. 1, No. 3, July, 1953.
5. Eargle, J.—"How Capacitor Mics Produce Cardioid Patterns," db Magazine, April, 1971.
6. K. Schoeps, Willy Kusters—U.S. Patent Nos. 2852620, 3190972, Karlsruhe-Durlach, Germany.

Albert B. Grundy
Director, Institute of Audio Research
New York, N.Y.

Response by Stephen F. Temmer, president of Gotham Audio Corporation.

The ads referred to were all about Neumann microphones and were strictly directed toward helping Neumann clients to select the proper model from among our twelve different ones. While Mr. Grundy's remarks about the microphones which he had represented for some years are true, we had never really felt that the word *switchable* referred to the positioning of a mechanical lever. We will in future use the adjective *electrically switchable* to avoid confusion.

It might also be interesting to note that the mechanically switchable pattern capsule was also invented by the same Mr. H. Grosskopf of the Institute for Broadcast Technology (I.R.T.) under U.S. Patent #2,787,671, who also invented the electrically switchable directional pattern. The Grosskopf patent referred to above was assigned to Schoeps, while the electrically switchable patent (2,678,967) was reassigned to Neumann by NWDR. The Kusters patent referred to by Mr. Grundy outlined only design and operating improvements on the basic Grosskopf patent.

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John M. Woram

THE SYNC TRACK

● Lately, there has been a series of letters asking for information on setting up small studios. One person inquired about books that discuss in depth the subject of building a small studio and, are there any companies that cater to the small studio population?

Well, there are no books, that I know of, on *How to Build a Small Studio,* or even, *How to Build a Large Studio* for that matter. Unfortunately, building a studio is not as clear cut as adding an outdoor patio or putting up a dormer in the attic. So, although there are published instructions for patio or dormer, no one—so far—has come up with studio plans.

Probably no one ever will. Or if they do, the book should be regarded with as much suspicion as the one that tells you how to mic drums. (see June *Sync Track*). Studio planning, and construction, depends on too many variables to be treated in text book format. Of course, general acoustic principles have been well covered in several texts (see February *Sync Track*), but the application of these principles to your studio cannot come from a book. Your personal requirements, and budget, will greatly influence the construction of your studio, and if you are unsure how to proceed, you should call in outside help before going ahead.

As for companies that cater to the small studio, there are of course many manufacturers of small consoles who may be consulted. When it comes to other equipment—microphones, speakers, signal processing devices, etc., the difference between small and large studios is primarily one of quantity, although the small operation may have to make do with more general-purpose type of gear, and leave the specialty items until later.

Naturally, the dealer would prefer a large account to a small one. Since the small studio population doesn't buy much, perhaps it doesn't get the attentive catering it needs. It's ironic, I suppose. The big time studio will have an experienced chief engineer who knows the business thoroughly. (Or at least he thinks he does). He doesn't need anyone's help or advice in choosing, and purchasing, equipment. Yet he gets more attention than he can stand because the supplier knows he's a "big spender." Down the block, the small studio operator who *needs* help can't get the time of day from his supplier because his account isn't worth the bother. I have many poignant memories of trying to get help in solving studio problems. Unless there was a definite possibility of a lot of money changing hands, I was strictly on my own.

Before all those wonderful "mom and pop" small guy-oriented shops start coming out of the woodwork and writing angry letters, let me point out that there is definitely another side to this coin.

Since getting involved in the consulting racket, I've gotten periodically depressed by the number of would-be small studio operators who are looking for free advice, wherever they can get it. Then, after they've gotten it, it's off to the nearest bargain basement discount house to buy their equipment. I suspect that the response I get to my ad is fairly typical of anyone in the same type of business. Via mail and phone come the inquiries: How do I do this or that? What kind of console do I need? What microphones

do I need? And on and on. Well of course I don't want to turn this kind of response off, but since I don't sell equipment directly, it would be very easy to go "down the tube" dispensing free information to all. So-o o, we turn delicately to the subject of $$$. A "consultation fee" it's called, and it is one of the most effective ways known to stop a conversation. The idea of paying for a service is just too much to bear. (Fortunately, not everyone feels this way, or I'd really be in trouble).

But anyway, you get the idea maybe? These same people will go out and bargain shop—going from store to store looking for the cheapest price they can get. I encourage my bride to do this—at the supermarket. If tomatoes are cheaper at the other store, she gets them there. And so on, down the grocery list. The economic advantages are obvious, but it is strictly up to my lady to find, and bring home, the bargains. Once she leaves the store the owner's interest vanishes. He knows damn well she will not be back unless his potatoes are on sale. And later on, if some of the food doesn't cook well, *he* certainly doesn't want to hear about it.

Now, if we can find our way out of the kitchen and back to the studio, what do you suppose happens when this bargain basement approach is tried when putting a small studio together? I'm talking, of course, to the point of view of the man who needs professional assistance.

What else? You wind up with a lot of problems. And, when you complain that the limiter won't limit, the dealer tells you it must be because of the console you bought (from a different dealer, naturally). And the console man blames it on the studio wiring job, which was done by the local high school shop class as a term project.

In short, nobody wants to know about your problems. And for good reasons too. Even if your limiter salesman was Mr. Sincere, he couldn't possibly figure out the mess in which his limiter has been placed. He knows that if he comes around to help, he's going to wind up inheriting all your other problems too, since he can't verify the limiter's performance if the rest of the system isn't working properly.

Where is all this leading?

If you're looking for a book on studio planning, it's because you don't feel qualified to tackle the job on your own. But—to continue the June discussion on books—you can't expect a book to do your planning for you. If you need help, get it. I presume the various letter writers need help, otherwise they wouldn't have written.

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In response to the increasingly complex needs of the audio community, the industry standard Model 550A, perhaps the most popular equalizer in the world, is now joined by a group of new devices providing extraordinary flexibility.

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Consult your distributor or the factory for specific data sheets.

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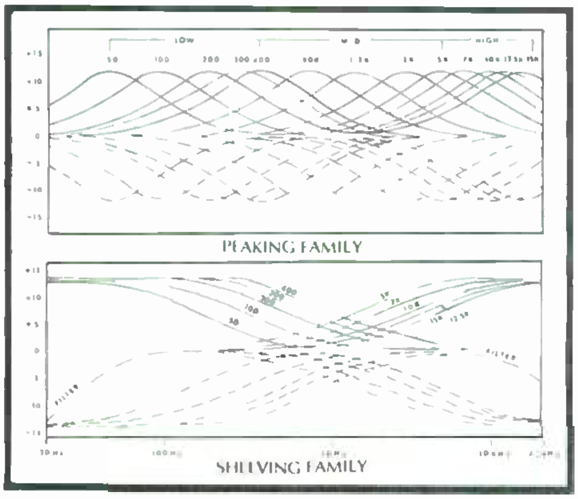

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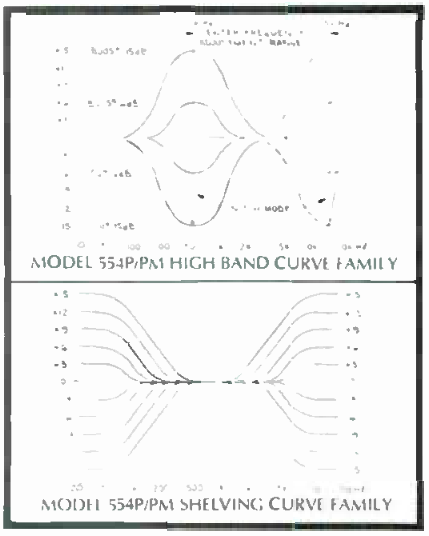
SHELVING FAMILY

NEW-MODEL 554P/PM

Dual Parametric manual and automated, continuously variable reciprocal peaking curves to a maximum of 15 dB boost or attenuation. A narrow "Q" notch is also provided, as are reciprocal shelving curves. The Model 554P provides flexibility and versatility consistent with the most advanced automated and manual audio mixing systems.



MODEL 554P/PM HIGH BAND CURVE FAMILY

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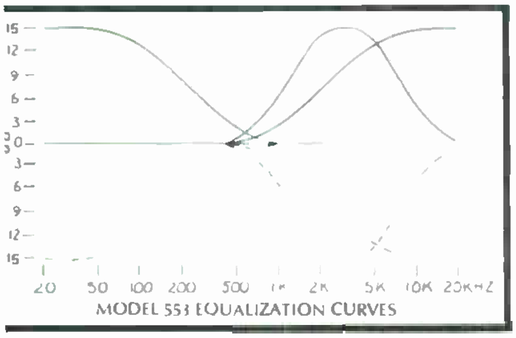
low cost, modular Equalizer suitable for a ide variety of applications in broadcasting, ·cording, film mixing, and sound ·inforcement installations. The shelving pe low and high frequency families of ırves produce familiar overall balance ıanges in the musical spectrum. The 3 kHz id-frequency peaking curves specifically fect the "presence" range of music and alogue.



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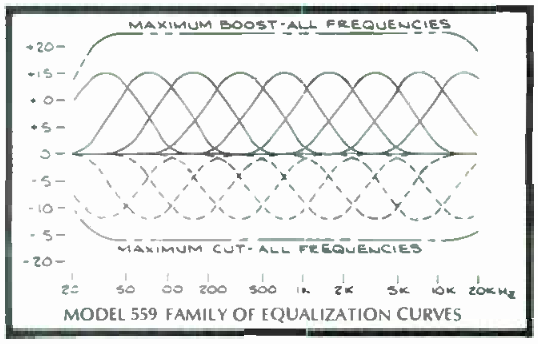
NEW-APSI MODEL 559

Simultaneous equalization in nine bands distributed over the three decades of the audio spectrum, with band centers at 35 Hz, 75 Hz, 160 Hz, 350 Hz, 750 Hz, 1600 Hz, 3500 Hz, 7500 Hz, and 16 kHz. Each band is controlled by a lever switch with an amplitude readout and provides reciprocal cut and boos of 2, 4, 6, 9, and 12 dB from unity gain and an additional 15 dB boost position.



MODEL 559 FAMILY OF EQUALIZATION CURVES

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What kind of help is available? Well, if you go to a responsible dealer, he'll be able to help you plan your purchases wisely. Don't be surprised though if he only recommends brands that he himself carries.

An independent consultant can be more unbiased in helping you, since he has no sales obligations, and can therefore recommend the equipment you need, and not what any one dealer would like to sell. But, he will expect to be compensated for his service, and if you can't face that prospect, a consultant is not for you.

Some dealers like to call themselves consultants. Some even keep a straight face while doing so. But as long as you're not being charged for the "consultation" you can humor him along. If he wants to charge you for telling you what to buy from him, beat it to the nearest exit.

Of course, if you need help much beyond planning what equipment to buy, then your dealer may ask to be compensated for additional services rendered. Which brings up an interesting point: how much service can you expect from your dealer in return for your business?

Historians tell us that many years ago, dealers would buy equipment wholesale, and resell it at retail. The dealer would make a profit, and would be genuinely glad to see you, since he valued all customers. Then something happened. Who knows where it started, but people started looking for discounts, not for quantity purchases, but just for the privilege of having the customer's business. As the discount syndrome took hold, store keepers searched for ways to cut corners, and one of the first things to go was service. Now, you walk in, haggle the price down as low as you can, and walk out with a "bargain." If you have trouble later, you know what you can do about it? Probably nothing—but you got what you paid for, didn't you?

Now, what with tight money and all that, some dealers are rethinking their discount policies. Some few companies have fanatically held to their list prices, and dealers who discounted have lost that company's product line. Although I haven't made a study of the subject, it seems no coincidence that these companies usually offer conspicuously better service too.

So, if you need help in planning your studio, never mind the bargain hunting. Find someone with the expertise to help you, and don't have a coronary if you are asked to pay for that help. If you go to a dealer who offers bargain basement prices, make sure your eyes are wide open, and find out in advance what sort of service to expect. If you know exactly what you're doing, then shop around for the best price, but don't expect attentive service later on. If you are not sure of your needs, or if you will require continuing help in getting your studio together, keep this in mind before spending your money.

As Julius Caesar said, "Caveat emptor!" (Or was it Orange Julius?)

COMING EVENTS

The 49th Audio Engineering Society Convention and Exhibition begins on September 9 and continues through September 12th in Fun City's plush Waldorf Astoria Hotel. There are details elsewhere in this issue on exact dates and times, but I want to draw your attention to four Educational Seminars that will be held during the Convention. These will be discussion groups by panels and not lectures. The four titles are 1. Introduction to Computer Programming, 2. Applications of the Desk Top Computer to Audio, 3. Tape Recorder Equalization, and 4. Practical Acoustics.

I want to emphasize the *practical* nature of these seminars. They should be of great value to working engineers attending the Convention. ■

Norman H. Crowhurst

THEORY AND PRACTICE

• A curious fact about this column, which alternates between discussions of education and the technical matters that really should be its business, and are the business of **db**, is that the content of my mail is usually based on my educational columns rather than those concerned with technical matters. And most of the comments are favorable. So **db**, the sound engineering magazine, has provided a platform for views that are not often expressed in the several hundred vehicles ostensibly devoted to educational material, but which too often subscribe to the apparently universal compulsion that professional educators share, to impress both their colleagues and the rest of the world with their erudition rather than their understanding of what is actually going on.

But this month we are going to discuss technical business. In an earlier column, we stated the difference between the kinds of distortion that the SMPE (now SMPTE) and CCIF methods were designed to measure. The SMPE method was designed to measure the kind of distortion that happens when large amplitude, low frequency waves change the gain, or amplification, offered to higher frequency, smaller amplitude waves, thus modulating them.

Back in the days when the method of measurement was devised, feedback was just about finding its way on the scene. It was so new that little theory about it had been published, and those who worked on the kind of distortion SMPE was then interested in were pursuing a different line of investigation from those who worked on feedback.

So, in that context, the main cause of that kind of distortion was the variation in gain that modulated the amplitude of the higher frequency components, at the point of the low frequency high amplitude wave. If there was any accompanying phase modulation, it was of no consequence, relatively speaking.

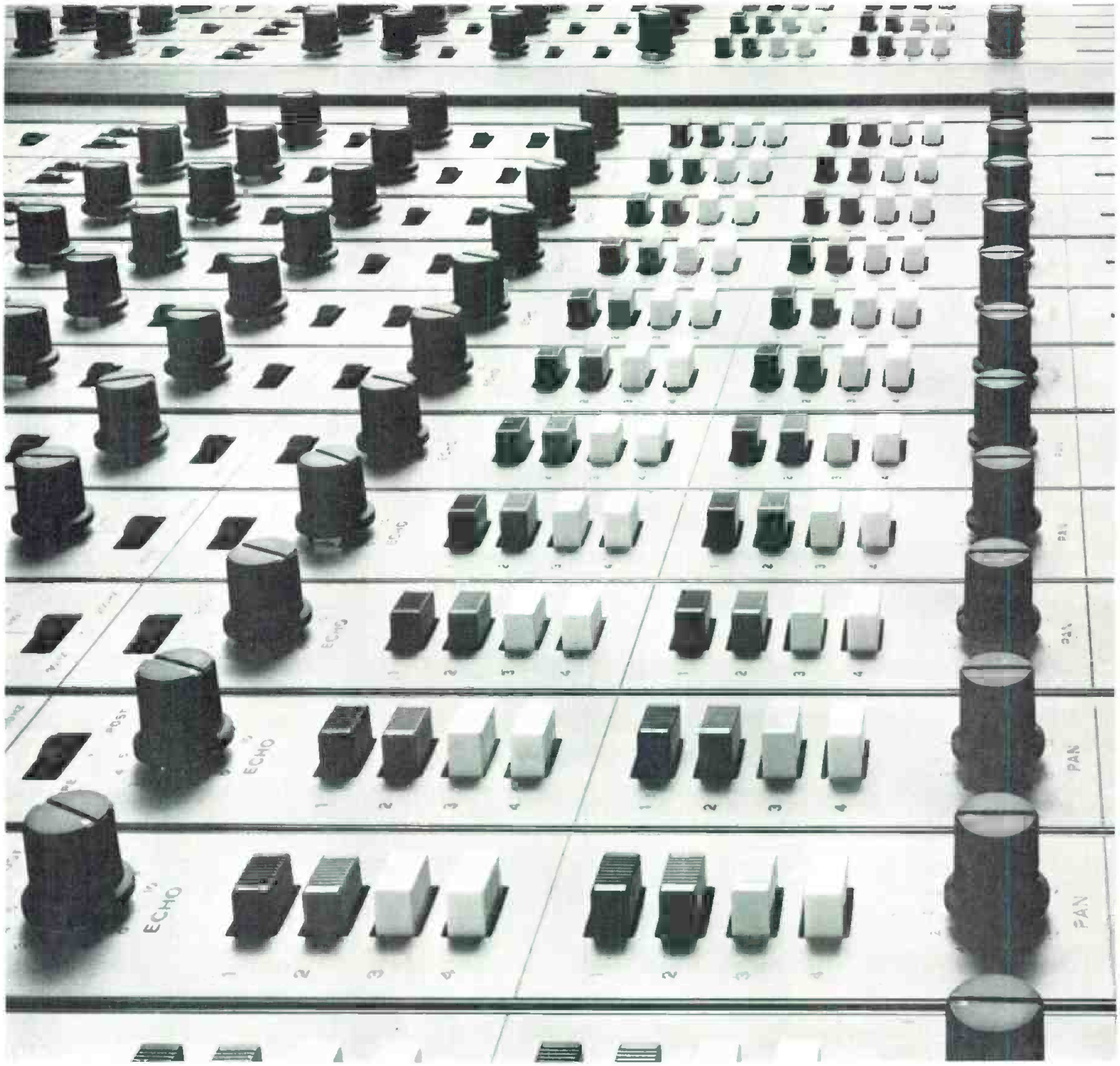
As everyone knows by now, one of the basic purposes of feedback is to reduce distortion, which includes the SMPTF type, along with everything else that qualifies as distortion. And as most people know, to apply feedback at all, phase shifts must be kept small; feedback will reduce them to something even smaller.

However, something that has been overlooked, the fact that feedback can produce SMPTE type intermodulation distortion, of the phase modulated variety. Over different parts, particularly of a large amplitude waveform, the dynamic resistance (also known as a.c. resistance) of active elements, whether they be tubes or transistors, changes over quite a wide range even when, as far as waveform is concerned, they are behaving linearly.

Perhaps I should explain that. When a tube or transistor is nearly cut off, its plate or collector resistance is many times the value it has at a moment when it approaches saturation. However, the voltages and currents associated with the waveform it amplifies may be very close to linear while this change in a.c. resistance occurs.

All these a.c. resistance values are associated with circuit reactances, either coupling capacitors, or stray circuit capacitance. And reactances produce phase shifts in the circuits where they are. When the resistance values change, the phase shifts change, although the waveform associated may not be distorted in any way. Such phase shifts that accompany the low frequency, high amplitude wave that produces the big swing will occur at any higher frequencies the same equipment is momentarily handling.

Now let's put it together. We may think that only amplitude modulation really matters. But this is not true. Consider a classic example, the difference between tremolo and vibrato.



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Now I'd better be careful because people from different backgrounds have different definitions for these musical qualities. The predominantly engineering definition, which I am using here, expresses *tremolo* as a fluctuation in amplitude, while *vibrato* is a fluctuation in frequency.

Now, as is done by organ manufacturers all the time in the ranks of stops they group together as celeste, this kind of effect can be produced by using two ranks of pipes, or signal generators, each having a slight difference in tuning, so they produce a beat.

Imagine the composite sound produced by two frequencies close together, like organ pipes, coming from a loudspeaker which radiates into free space, so you are not troubled with standing waves to complicate matters. Because you have two frequencies, you will have two corresponding wavelengths, which means that at no two points in space will the same combination occur as you vary your distance from the source.

For this reason, at some points the result will be almost pure amplitude fluctuation while at others it will be almost pure phase, or equivalent frequency fluctuation. This is why you have such a hard time telling the difference between tremolo and vibrato. While each can produce its own effect to an extent that makes it not convertible into the other, they often seem to become equivalent to each other at certain points out in space where you do your listening.

From the causative point of view, of course, there are differences. If you use a variable gain stage to modulate the audio, the basic effect has to be tremolo. If you feed the audio to a loudspeaker that is somehow waved around in space, the basic effect has to be vibrato. And of course, the effect that Paul Klipsch has done so much to expose, as a deficiency of small-diaphragm wide-range speakers that he calls *doppler* distortion, is an unwanted frequency or phase modulation.

The point we started out to make here is that the SMPTE method does not, basically, detect that variety, although its effect on what you hear can be precisely similar to the variety the SMPTE method can detect. This is because the method of measurement consists of mixing the two frequencies together, a large low one with a small high one, feeding it through the equipment to be measured, and then analyzing the output in the following way.

First it goes through a high pass filter, that removes the low frequency component. Then the remaining high

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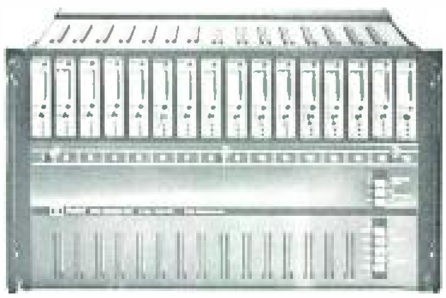
Professional Recording and Transmission Applications

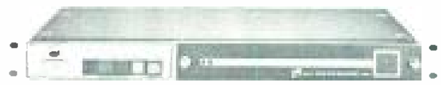
360
The Dolby 360 is a basic single-channel A-type noise reduction unit for encoding or decoding. This unit is normally used in a fixed mode such as in disc cutting or landline sending or receiving, the operating mode is manually selected.

361
The Dolby 361 is similar to the 360, providing a single channel of A-type noise reduction, but with relay switching of operating mode and tape recorder connections. The changeover can be controlled automatically by the recorder.



M-Series
The Dolby M16 A-type unit is designed specifically for professional multi-track recording, and incorporates 16 channels of noise reduction in a compact chassis only 10 ¼ inches high. The similar M8 is an 8-track version, and the M8X allows simple extension of the M16 for 24-track use.

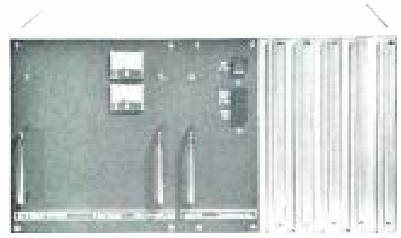
Motion Picture Industry



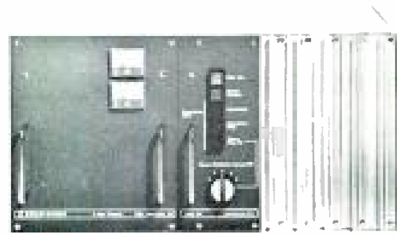
364
The Dolby 364 Cinema Noise Reduction Unit is intended primarily for use with Dolby A-type encoded optical sound-tracks. The 364 also includes a standard 'Academy' filter for conventional tracks, a clean-up circuit for old or worn prints, and provision for playback of magnetic sound-tracks with or without Dolby system encoding.

E2
The Dolby E2 Cinema Equalizer is a companion unit to the 364, and has been specifically designed to solve the response equalization problems of cinemas. Used with the 364 and Dolbyized optical sound-tracks, the E2 enables most cinemas to achieve modern sound reproduction standards without replacement of existing equipment.

Professional Encoders for Consumer Media



320
The Dolby 320 Duplication Processor is a professional quality unit with B-type (consumer) noise reduction characteristics. The unit is used for encoding duplication master tapes in the high-speed duplication of Dolbyized cassettes, cartridges, and open-reel tapes. The 320 is a two-channel unit.



324
The 324 Broadcast Encoder allows broadcast stations to encode stereo FM broadcasts with the Dolby B-type characteristic. The unit provides for an optional reduction of high frequency pre-emphasis, reducing the need for high frequency compression, and thus allowing a significant additional improvement of reception quality.

Noise Reduction Module

Cat 22
The Dolby noise reduction module, Cat 22, is the basic functional unit employed in all A-type equipment. The Cat 22 is available as a spare or in quantity to OEM users for factory installation. A half-speed version of the module (Cat 40) is also available.

Test Set (A-type)

Cat 35
The Dolby NRM Test Set, Cat 35, permits rapid verification of performance of Cat 22 Noise Reduction Modules without their removal or the need for additional test equipment.

Dolby™

For detailed information contact
Dolby Laboratories Inc
1133 Avenue of the Americas
New York NY 10036
Telephone (212) 489-6652
Telex 125797

346 Clapham Road
London SW9
Telephone 01-720 1111
Telex 919109

Circle 17 on Reader Service Card

frequency component is rectified and filtered to find whether its amplitude has any low frequency fluctuation. What this does not do is to determine whether its frequency or phase fluctuates. The rectification and filtering could not care whether frequency changes or not.

Before passing to the CCIF method and its deficiencies in measurement, we need to show that the two forms do not measure equivalent forms of distortion. The fact we mentioned a little earlier, that two higher frequencies close together are used to produce the celeste effect, as well as other beat effects, has led many to believe that the higher notes produce a lower note.

This is not true. The beat is not itself a note. In the first place, the beat is too slow to be audible as a note. But if you increase the difference between the frequencies, so the difference could be a note, it is not, as long as there is no distortion present, able to produce the CCIF type of byproduct.

The big difference is in cause and effect. With the SMPTE method, we start with a low frequency and a high frequency. The equipment must handle both. If, in doing so, one affects the other, you have SMPTE distortion. With the CCIF method, we start with two high frequencies. For this purpose, the equipment need not even handle the low, distortion-product frequency. So if you applied the SMPTE test to it, nothing would show, because the low frequency would not even enter the system.

The CCIF method produces the low frequency as a distortion product, that should not be there, and was not present in the input.

Now let us take a closer look at the CCIF variety of distortion. Suppose the two frequencies are 5000 Hz and 5100 Hz. If the system has asymmetric distortion, such as is due to an unbalanced push-pull stage somewhere, then the beat produced by these two frequencies, at 100 Hz, will have bigger tops than bottoms on the wave envelope produced. This will mean that, as well as the 100 Hz modulation envelope of the high frequency waves, there will be a 100 Hz component in the signal. This will appear, audible, as a 100 Hz "buzz."

That is what the CCIF method is designed to detect. But now, suppose the distortion is not asymmetrical, but symmetrical. Now the envelope will have its shape either flattened at top and bottom or stretched at top and bottom. It is not so easy to see what the additional byproducts will be in this case as it was with the asymmetrical case. But mathematical analysis shows that this is equivalent to a whole new bag of extra frequencies, none of which will be anywhere nearly as low as a few hundred Hz.

By a different choice of test frequencies, you can deliberately produce byproducts wherever you want, and the test can perform more like a waveform analyzer does in taking harmonic distortion apart. But to date, to our knowledge, no such method of measurement has been formalized. So all the CCIF method really detects is the kind of unbalance in a system that produces first-order, asymmetric distortion.

One of my recent correspondents reversed the usual way in which I transfer things, which is to show that what happens in education is like what happens among audio engineers. I had said, in a column I write for general audience readership, that the reason professional educators clamp down on anyone who attempts to say something unconventional, is that they are afraid if they allow it, the teachers will soon be out of their depth.

This man's comment said that he has found the same thing in engineering circles. That any suggestion that breaks with established convention makes engineers nervous too, because they feel they will soon be out of their depth. You know, I think he's right. Why else hasn't someone developed more sophisticated tests, so that distortion can be more exactly analyzed, more easily?

Probably because most engineers who use existing methods, do not really know what those measurements tell them, and if anyone suggests using a more sophisticated kind of test, with which they are unfamiliar, they fear they will soon be confused.

My attitude through life has always been to accept something strange as a challenge, not to reject it out of hand. I want to find out how it relates to what I already know, or think I know. And invariably accepting of the challenge leads to better understanding of something I would not have expected. Why do so many of us go around with mental blinkers on, afraid we might discover something useful? ■

Circle 32 on Reader Service Card

Circle 28 on Reader Service Card

Last year, we offered you Studer's tape recorders.

This year, Telefunken's.

What happened?

The answer is on page 3 of our catalogue.

In the "Statement of Principles" bi-annually published there — exactly as we wrote it seventeen years ago.

Since then, we've always vowed that, before we changed a word of that credo, we'd rather see the list of our companies change.

And this has now happened.

As our credo makes clear, Gotham prefers to represent "equipment made in limited quantity for a discriminating worldwide clientele."

This was Studer's orientation when we joined forces with them many years ago. A tiny company — only a handful of people — dedicated to serving a *limited* professional market.

But no more. Over the years they have expanded enormously. Requiring mass marketing goals.

It is a perfectly legitimate development we can respect. But. . .

The credo, remember.

Ah, you say. But the credo also stresses Gotham's interest in representing companies that are personally-run by men who *"devote their lives to the fulfillment of their dreams."* Surely, you say, the gigantic Telefunken operation can hardly be described as personally-run.

No, it can't.

But the tiny enclave that we represent *can*.

For, within Telefunken there is a separate department of about 45 people. Autonomous, because they are completely apart — independently handling *all* the development, manufacturing, and marketing of the M12 and M15 "Magnetophon" professional recorders.

These people have a high and unique responsibility — because it was Telefunken, after all, who invented the tape recorder in 1942.

And how they do their job conforms precisely with our credo. Because, to them as to Gotham: *"Not price but excellence, craftsmanship and serviceability are the criteria"* that count.

Some reasons why the M12 "Magnetophon" is in a class by itself:

1. A 19" rack mountable 10½" reel ¼" tape recorder incorporating an optional 4-input mike/line slide fader mixer. Mike inputs are Phantom® powered, of course, ready for Neumann fet 80 microphones.
2. Heads made of ferrite and sintered ruby tape guides; materials so smooth and hard we guarantee them both for the life of the machine (15 years).
3. Ingenious tape tension control on *both* sides of the capstan; no reel size switching.
4. Weighted Peak Flutter typically ± 0.02% at 15 ips; 3¾ ips speed available.
5. Price range: $3500 to $4500.

And after the technicalities, there is also the matter of experience:

For over five years now, our experience with the "Magnetophon" in Neumann's tape-to-disk transfer systems has been eminently satisfying.

Satisfying? Superb!

And we look forward to the same kind of results with Telefunken's M12 and M15 "Magnetophon." For ourselves and for you.

Write today for more information:

TELE FUN KEN

GOTHAM
AUDIO CORPORATION

Headquarters:
741 Washington St., New York, NY 10014
(212) 741-7411

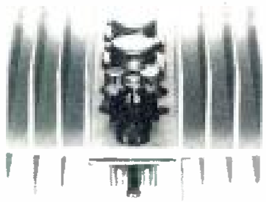
West Coast Sales Office:
1710 N. LaBrea Ave., Hollywood, CA 90046
(213) 874-4444

NEW PRODUCTS AND SERVICES

FOUR-WAY MONITOR

● Model 4340/41 four-way monitor is designed for bi-amplification use with the manufacturer's 5231/32 electronic crossover or with high level passive crossover networks. The system contains a fifteen-inch low frequency loudspeaker, ten-inch midrange loudspeaker, high frequency compression driver, and ultra high frequency compression driver, and has a frequency response of 35 to 20,000 Hz +3 dB. Sensitivity (SPL at 30 ft., 1 mW) is 44 dB. Power output (SPL at 10 ft. in a room volume of 2,000 cu. ft. with 1/.2 rated power input, 35.5 watts), is 101 dB. The crossover frequency is 250, 1,250 and 9,500 Hz.
Mfr: James B. Lansing Sound, Inc.
Circle 48 on Reader Service Card

OPERATIONAL POWER AMPLIFIER

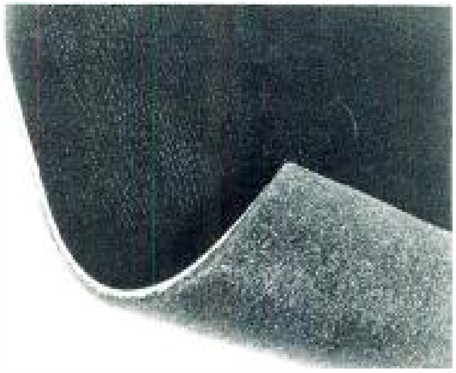


● Intended to be used as a servo motor or d.c. through audio power amplifier, model 440 differential d.c. operational power amplifier consists of an Opamp 4009 driving a dual class AB power amplifier. The manufacturer claims there is no crossover distortion. The unit may be used in audio applications with either a single polarity or bi-polar power supply. It has an output capability of 50 watts rms and is constructed on the octal plug-in heat sink, with the circuitry isolated from the case. Available either as a kit or wired and tested.
Mfr: Opamp Labs, Inc.
Price: Kit: $35; Wired & Tested: $60.00
Circle 49 on Reader Service Card

DYNAMIC SAFETY MICROPHONE

● Flexible rubber housing offers maximum security for dynamic unidirectional moving coil microphone M412, a feature of interest for field use. The microphone has a tailored acoustical characteristic especially suitable for close talking and can be gooseneck mounted or hand held. It has a built-in switch for voice on/off or for relay actuation. The frequency response is 200-12,000 Hz.
Mfr: Revox (Beyer-Dynamic)
Price: $75
Circle 50 on Reader Service Card

SOUND ABSORBER & VIBRATION ISOLATOR



● Sound barrier Soundmat FV is composed of a one-pound per square foot of limp mass, acting as the barrier layer, which is bonded to ¼ inch layer of acoustic foam. The acoustic foam is fire retardant. The material can be applied to floors or around resonating pipes by die-cutting and shaping to fit. For exposed areas, a heavy black vinyl skin can be laminated to the barrier, providing a scuff-resistant, water-barrier surface.
Mfr: Soundcoat
Circle 53 on Reader Service Card

TAPE CONTROL

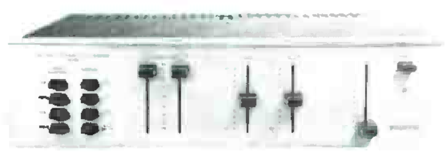
● Constant tension from beginning to end of any size tape reel is achieved by controlling the torque of the supply reel motor by the Tentrol controller. The device eliminates tape speed changes, and thus changes in pitch, with tape reels up to 14 inches. An additional unit can be installed on the takeup motor to provide constant takeup tension as well. Two units can provide fast starts at 30 ips, eliminating capstan creep. The unit is designed to fit nearly all Ampex tape recorders, as well as other well known machines.
Mfr: Inovonics
Circle 54 on Reader Service Card

VARIABLE SPEED OSCILLATOR

● Available with an optional four-digit electronic display for exact speed accuracy, the lightweight VS-10 is specifically designed for use with the Ampex MM-1100 and AG-440 series of recorder/reproducers and can drive up to three recorders. The self-contained oscillator, powered by the capstan servo of the recorder via a single cable, features a range of ± full tone in quarter-tone steps and a coarse/fine variable speed adjustment. The readout display (110/220 vac, 50/60 Hz) may be operated either in percent of nominal speed or in frequency (50 or 60 Hz center frequency) of the associated recorder. The readout utilizes the servo's crystal reference for control.

Mfr: Ampex Corporation
Price: $795 (with optional digital readout display) $395 without.
Circle 51 on Reader Service Card

HIGH DEFINITION STEREO AMPLIFIER AND PREAMPLIFIER

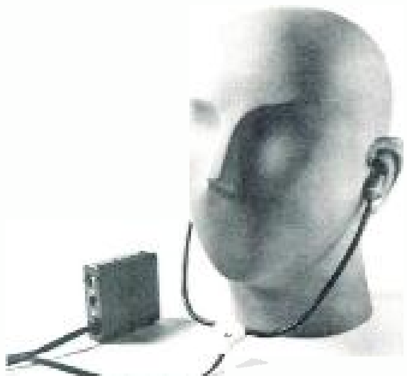


● New circuitry concepts, which it is claimed produces an effect with no discernable distortion apparent below the actual noise of the circuit, are used in the preamplifier design of HD250 integrated control amplifier and HD22 high definition preamplifier. With the tone controls in the flat position and measured over a bandwidth from 15 Hz to 23 kHz, the signal-to-noise ratio with respect to an input figure of 5mV is −83dB. The amplifier section is capable of producing 50 watts per channel at 8 ohms, both channels driven, from 20 Hz to 20 kHz ± 1 dB, with an average of 90 watts per channel at 5-6 ohms.

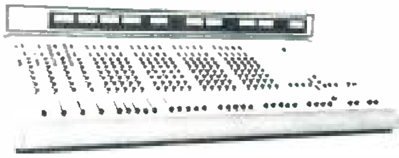
Mfr: Audionics
Price: Amplifier: $695
Preamplifier: $449
Circle 52 on Reader Service Card

BINAURAL MICROPHONE

● Physical listening conditions are simulated with improved fidelity, according to the manufacturer, by the use of MKE 2002, a binaural microphone which fits over the recordist's ears. The unit, mounted on a stethoscope-like band over the recordist's head, contains two electret transducers which rest lightly in the ears. This permits a reproduction of all the variables present in human hearing, the structure of the head, surrounding air noise, left-and-rightness, the perception of distance in addition to differences in tone, etc. The result is not only a lifelike pickup of a performance, but also the surrounding audio ambience that would be present in an audience situation. The dual condenser microphones come complete with battery power supply, carrying case, and plastic artificial head, willing to stand in for the actual recordist if necessary.
Mfr: Sennheiser
Price: $330
Circle 45 on Reader Service Card

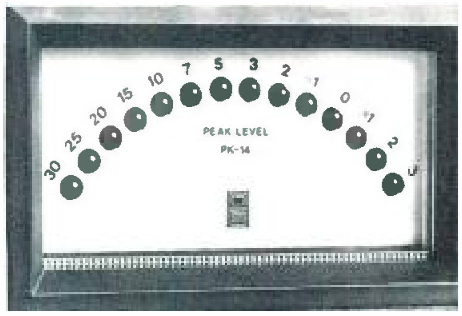


AUDIO CONTROL CONSOLE



● Input module functions present in series 20 LM console include four inputs, four pre-settable mute controls, phase switching mic and line switches and attenuation, three-knob eq., stereo pan pot, all with in/out switching. Two foldback and an echo send controls are provided with in/out switching and pre-post fader selection. The unit has conductive plastic slide attenuation, monitor solo function and on/off switch with led indicator. The stock model includes sixteen inputs (expandable to 32 inputs), quad outputs with a mono bus, stereo buses, two foldback, and one echo send bus. All ten outputs are metered. A computer ribbon cable organization permits input and output panels to be located in any board position. Additional options can be added.
Mfr: Cétec
Price: $11,500 (additional input modules $375 each)
Circle 46 on Reader Service Card

PEAK PROGRAM METERS



● All solid-state peak program meters PK-14 and PK-16 offer precise visual level indication via sequentially lighted led readouts of true signal levels. A 14 or 16 green and red segmented readout will monitor standard vu or peak level program content, depending upon the user's preference in selecting its alternate mode operation by a front panel control setting. Both units contain led brightness control, integration and decay time adjustment, input level scaling, constant current consumption, simple power supply requirements. Integration time specifications conform to European DIN 45406 for peak reading and USA C16.5—1954 for vu characteristics. The PK-14 arc-scale meter is designed to retrofit into existing 3½ inch meter installations, while the vertical scale PK-16 offers monitoring capabilities in updated or custom designed control consoles.
Mfr: Quad/Eight Electronics
Circle 44 on Reader Service Card

"ROCK MONITOR" AIR MOTION LOUDSPEAKER

● The requirements of modern musical forms are met by AMT 3 loudspeaker, which the manufacturer claims to offer support to the transient performance obtainable from a conventional cone bass system. Midrange and treble frequencies are handled by an air-motion transformer positioned for unhindered radiation in all directions. For lower frequencies, a six inch high compliance, high gauss midrange driver is mounted in a resistively terminated line which extends back into the bass enclosure. The lowest frequencies are served by a proper matching of the midrange to twin ten inch drivers. A carefully calculated resistance allows the drivers to continue linear operation far below system resonance and without uncontrolled motion at any frequency.
Mfr: Ess, Inc. (Heil)
Price: $435
Circle 47 on Reader Service Card

DIGITAL TIMER CLOCK

• Both a bright visual display of correct time and form "C" contact closures are provided with model DC24 digital clock. It measures 19 inches by 3½ inches high by 10 inches deep. Options include reset board (approximate time corrections up to four times per hour), net join board (exact time corrections either once or twice per hour), and oscillator time base and battery backup board, which maintain the exact time independently of line frequency variations or power interruptions. Controls on the front panel enable the operator to halt the clock altogether, advance the seconds counter rapidly, advance the minutes counter rapidly, or light the led display memomentarily when operating the DC 24 on battery.

Mfr: Sparta Electronic Corp.
Price: (With reset board) $475. Net join option: $100 Oscillator/battery backup option: $125.
Circle 41 on Reader Service Card

QUADRIPHONIC SYSTEM CONTROL CENTER

• Designed for true discrete systems of from one to four channels, model 4xPA preamplifier offers a built-in CD-4 demodulator, provisions for SQ demodulator, accommodation for 2. 4 channel tape recorders, an equalizer filter system, and a 4 x 4 matrix mode control allowing any input channel to be assigned to any output channel. The frequency response is, at high level, ± 0.1 dB, 20 Hz to 20 kHz at rated output, down −3 dB at 1 Hz and 100 kHz at rated output. The phono range is ± 0.25 dB from RIAA curve at rated output.

Mfr: BGW Systems
Price: $849
Circle 43 on Reader Service Card

AUDIO FLUTTER METER

• Designed for field operations, model 8160 portable flutter meter complies with the IEEE and DIN recommended standards, for pulse response. In addition to outside work, it can be used in hi-fi and radio repair shops, audio and video tape transports production line tests, and for servicing dictating equipment.

Mfr: 3M
Mfr: $395
Circle 42 on Reader Service Card

Martin Dickstein

SOUND WITH IMAGES

Equipment Follow-Up

● A short time ago, we devoted a column to the care and feeding of the 16mm projector. In a subsequent column we quoted a letter we received from a reader who shared his experiences with maintaining and handling film projectors. Just a couple of months ago, this column discussed a specific film projector to introduce it to some of our readers who might not have been familiar with it. Once again, we are pleased that a reader took the time to write of his experiences with this unit, and to introduce us to another double system projector. We should like to follow up as quickly as possible this time, and then, after the letter, we'll provide some of the technical specifications for the Bauer Studio projector mentioned in comparison with the Siemans. Incidentally, sincere thanks to Mr. John M. Hoerner, Jr., Extension Editor-Visual Communications, University of Georgia College of Agriculture, Athens, Ga., for his interesting comments.

"The Siemans double sixteen projector was an excellent tool for single and double system film editing. I spent several years learning its idiosyncracies and capabilities, but it has now been replaced by the Bauer Studio projector. The mixing capability of the Siemans was often very useful, but without an external vu meter to be able to read the original track level, the overdub level was often too high or too low, and the lack of a monitor head made the process too hit or miss to be reliable for news film production. Rather than try an overdub mix, I would transfer the sound to the full coat side, and then do the mix coming back to the single system side and the sound track on the film.

"Maintaining sync when the film was removed for editing after the sound was transferred was often a problem because of the air damped idler on the full coat side. I always spliced a frame of beep tone into the leaders of the film before transfer, made the transfer, matched up the beeps on a syncronizer, then retarded the track 28 frames for editing. When the film and sound track were loaded back on the projector for transfer back to the mag. stripe, the cue punches were appropriately lined up. Then I would run the projector backwards for a few feet, and, without going into the record mode, I would run it forward and listen for the beeps. If they matched, then I would go into record and complete the transfer. I found this was the fastest and most accurate method of maintaining sync when editing.

"The removable gate of the Siemans

was a distinct advantage which was not present on the Bauer projector. During transfer operations, I would always remove the gate to prevent any additional wear on the film and to avoid the possibility of breaking a splice. If several transfer operations are anticipated, I would recommend splicing with clear Mylar tape, rather than hot splicing. Another problem created by the Siemans was electrical noise generated by a governor-controlled motor. While synchronous motors were available on the Siemans, I never knew anyone among a dozen or so machine owners who had one. The governor incorporated a set of points on the back of the motor shaft, and they would pit and burn, requiring dressing to minimize the electrical interference.

"Despite the few problems, the Siemans was an excellent projector, and one of the few I would trust to project original film without scratching. The pressure plate was made of a type of bakelite, with a floating metal plate, and a triple pulldown claw. Film could be threaded to bypass the sound head, further minimizing the chance of scratches.

"The Bauer studio projector will do everything the Siemans does, and has a few extra features, among them a synchronous motor which is standard, optional 4-channel heads, including a monitor head, and improved flutter and wow specifications. Also, once both sides of the projector are threaded, the mechanical interlock can be recoupled for syncing the tracks, while the Siemans had to be rethreaded."

We appreciate comments like those of Mr. Hoerner which come from repeated experience, and which he is willing to share with others. It is not the purpose of this column to compare directly two similar pieces of equipment, but to introduce these units to those to whom they may be unfamiliar. The following is, then, the specifications for the *Bauer P6* studio double band projector, and these will be followed by a brief note on another similar unit, the *Sonorex 16/16*.

The Bauer has an output of approximately 500 lumens, using a 250W quartz iodine lamp, and is cooled by a double fan on the motor shaft. The drive is by synchronous motor, the take-up assembly is load controlled and self compensating for uniform winding, there is provision for fast rewind, and an automatic fail-safe switch is incorporated to stop the projector if the film should break. The film moving mechanism is a 3-toothed claw, the film pull-down ratio is 1:6.9, and the picture steadiness is plus or minus 0.1 percent.

In the sound department, provision is made to have a monitoring facility during recording. A built-in speaker is provided with a level control switch, the power amplifier is rated at 25W music power with a frequency response of 50-7,000 Hz. (plus or minus 3 dB) for the optical sound and 50-12 kHz. (plus or minus 3 dB) for mag. sound. The signal-to-noise ratio is −45dB. Wow and flutter are specified at plus or minus 0.4 percent. The amplifier output can feed an external 8 ohm speaker, and a preamp output is also provided with a 1.5 volt output at 600 ohms. The unit also has an hour timer built in.

The Sonorex 16/16 is similar to the Bauer in several aspects. It is a double system unit and has a sync motor, double flywheels with motor-driven run-up for fast start, a safety switch for end-or-break of film, an elapsed time meter, and monitoring facilities.

If anyone out there in film-editing land has had other experiences with projectors of this or any other type, or if you are familiar with any other units in the 16mm double system, please let us have your comments for others to read in this column. ■

49th AES Convention and Exhibition

On these pages, we present the essential program and an exhibition map of the Audio Engineering Society's 49th Convention and Exhibition to be held at New York City's Waldorf-Astoria Hotel. The dates are September 9 through September 12, 1974.

Schedule of Events

EXHIBIT HOURS

Monday and Tuesday, September 9 and 10
1:00 P.M. to 9:00 P.M.
Wednesday and Thursday, September 11 and 12
11:00 A.M. to 5:00 P.M.

TECHNICAL SESSIONS

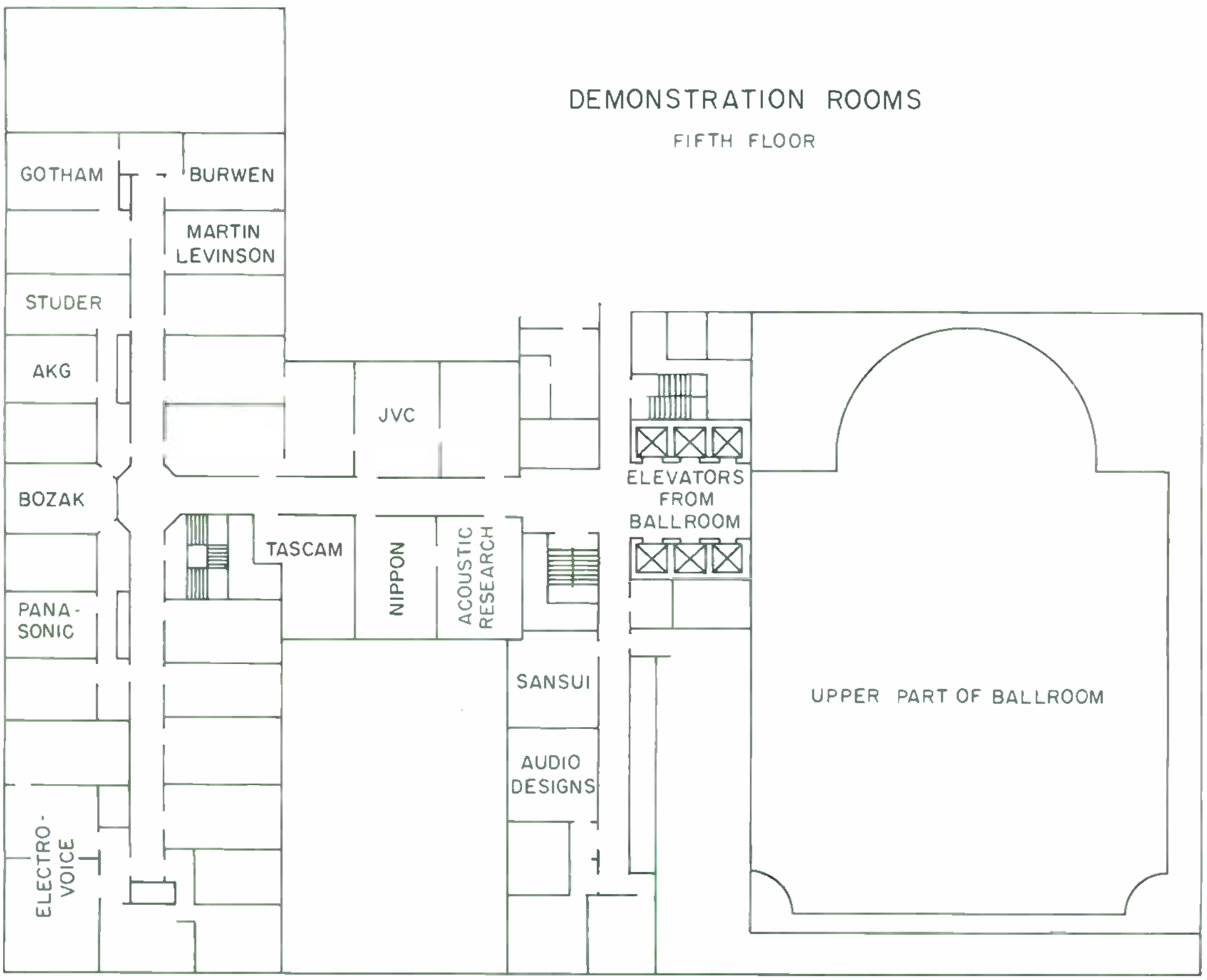
MONDAY, September 9
- 9:00 A.M.—Annual Business Meeting
- 9:30 A.M.—A—Audio in Broadcasting
- 9:30 A.M.—B—Audio in Medicine
- 2:30 P.M.—C—Architectural Acoustics
- 2:30 P.M.—D—Introduction to Computer Programming
- 7:30 P.M.—E—Broadcasting Music Recording Abroad
- 7:30 P.M.—F—Application of Desk-Top Computer to Audio

TUESDAY, September 10
- 9:30 A.M.—G—Transducers I
- 9:30 A.M.—H—Tape Recorder Equalization
- 2:30 P.M.—I—Transducers II
- 2:30 P.M.—J—Practical Acoustics
- 4:00 P.M.—K—Studio Design (Panel)
- 7:30 P.M.—L—New York Section presents Synchronous Sound Systems for Disneyland Parades

WEDNESDAY, September 11
- 9:30 A.M.—M—Signal Processing
- 2:30 P.M.—N—Magnetic Recording
- 4:00 P.M.—O—Forensic Audio Engineering: Application of Audio Engineering to Civil and Criminal Law
- 7:00 P.M.—Cocktail Party
- 8:00 P.M.—Awards Banquet

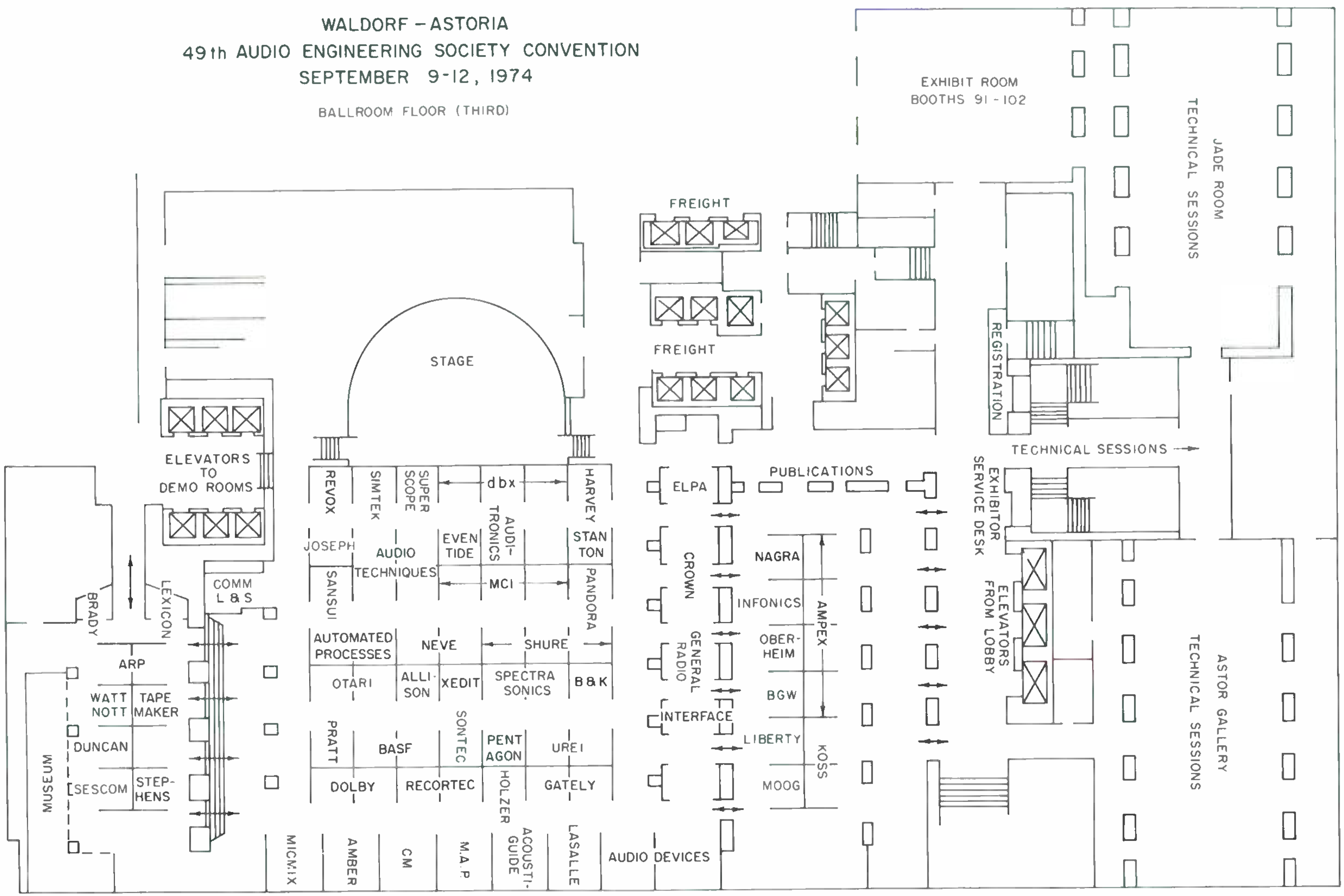
THURSDAY, September 12
- 9:30 A.M.—P—Disc Recording
- 2:30 P.M.—Q—Audio Instrumentation
- 2:30 P.M.—R—Electronic Music . . . Concert
- 7:30 P.M.—S—How Valid are Hi-Fi Equipment Tests? (Panel)

Exhibit areas will be the ballroom, available from the third floor with additional demonstrations being given in rooms on the fifth floor.



WALDORF - ASTORIA
49th AUDIO ENGINEERING SOCIETY CONVENTION
SEPTEMBER 9-12, 1974

BALLROOM FLOOR (THIRD)




Keep on Trackin'
Multi-trackin, that is, with Stephens Electronics professional tape recorder/reproducers. Our superbly designed and engineered machines are available in a variety of standard formats, from 4 to 24 tracks, or you can keep on trackin with Stephens' unique 32 track and 40 track 2-inch machines.
Pictured (front-to-rear) are the 4-track, 40-track and 24-track machines.
Battery powered portable units are also available.
STEPHENS ELECTRONICS, INC
3513 Pacific Ave., Burbank, Calif. 915[illegible] Phone (213) 842-5116
Circle 19 on Reader Service Card

SIDNEY L. SILVER

Controlled Time Delays for Speech Reinforcement Systems

Time delay applications can significantly improve the performance of a sound reinforcement system, particularly if it is used for speech improvement. The author covers such concepts in detail.

IN LARGE CONFERENCE ROOMS and assembly halls employing speech reinforcement systems, disturbing time lags are often produced when amplified speech from a nearby loudspeaker, or even a more distant loudspeaker, reaches the listener's ears well before the direct sound from a live source. The resulting sound may be perceived by the audience as discrete annoying echoes, or, when there is a profusion of echoes more closely spaced in time, as excessive reverberation. To compensate for these undesirable time lags, time-delay units are often introduced into the amplification system for the purpose of retarding the sound from a particular loudspeaker (or a grouping of loudspeakers), so that the total sound pattern is synchronized with the live sound source.

Time-delay correction, in effect, can help maintain the proper spatio-temporal relationship between natural and reinforced speech, permitting the human hearing mechanism to integrate subjectively, and hence localize the sound as coming from the *true* source. Depending upon the acoustical characteristics of the hall and the location and orientation of the loudspeaker system, time-delay units may be used to enhance the naturalness of speech, improve intelligibility, and provide a high degree of directional realism. Before discussing this subject in greater detail, however, let us examine some of the psychological and physical phenomena relating to the human auditory system, which create a demand for the practical application of these devices.

Sidney L. Silver, a frequent contributor to this publication, is with the United Nations Telecommunications Section where he is in charge of sound and recording.

PSYCHOACOUSTICAL DATA

It has long been recognized that when a number of separate sound sources (similar in content, and not too widely different in intensity) follow each other in close sequence, we attribute the total sound to that source leading in time. Subjective experiments by Haas[1] and other acousticians[2] have provided an adequate basis for understanding the psychoacoustic events involved in the human perception of speech sounds arriving from multiple transmission paths. The Haas effect, or *precedence* effect, as it is also called, has established that for successively arriving sounds with small differences in delay time, the earlier sound, by suppressing the audible effects of the later sounds, will dominate the spatial impression. For larger differences in delay time, the sounds will be heard as individual echoes, distinct both in space and time. This delay phenomenon, while primarily concerned with room acoustics, has had considerable influence on the design of speech reinforcement systems.

To demonstrate the precedence effect, two loudspeakers (connected in-phase) are placed in front of an observer in an anechoic environment. Using pre-recorded speech as the program source, the levels of both loudspeakers are adjusted for equal intensity. Referring to FIGURE 1, the primary loudspeaker represents the direct sound source and the secondary loudspeaker, a delayed version of the original sound. Here the secondary source is intended to simulate the effect of a single reflection of the direct sound under actual acoustical conditions. Relating this set-up to speech reinforcement systems, the primary sound source may be regarded as the talker, and the secondary source as the amplified speech signals.

Initially, with no delay on the secondary loudspeaker, the virtual sound image appears to emanate from a single source located at a *phantom* position between the two loudspeakers. As the sound from the secondary source is delayed (on the order of 0.5 msec), the sound image gradually shifts toward the primary loudspeaker, but the point source location is not well-defined and directional stability is rather poor. If now the time delay is increased to 5 msec, the direct sound will completely mask the delayed sound, and this effect will be maintained up to about 35 msec. Within this range of delays, the hearing mechanism is characterized by an auditory fusion period during which the primary and secondary sound stimuli seem to coalesce into a single, more intense stimulus. The ear-brain perceptual mechanism apparently fully integrates the acoustic energy from these sources in such a way that localization of the combined sound is determined by the direction of the original sound source.

As the delay is increased beyond 35 msec, the delayed sound is only partially integrated with the direct sound; the listener begins to perceive the secondary sound, but the initial sound is still localized at the primary source. Finally, at delays of about 50 msec or larger, there is a discontinuity of the sound signals and the delayed sound is heard as a separate echo of the direct sound. At this stage, listening becomes unpleasantly strained as both signals seem to arrive from different directions, thus detracting from the quality and interfering with the naturalness of speech.

In order to determine quantitatively the extent of the temporal masking process, the observer is then allowed to reduce the intensity of the direct sound as a function of varying amounts of time delay, so as to produce equal loudness from both primary and secondary loudspeakers. Under these conditions, the directional sense becomes obscured and the listener cannot distinguish which of the loudspeakers is the original source. From the curve plotted in FIGURE 2 (representing the average impressions of a group of observers), it can be seen that point source location can be maintained within most of the delay range between 5 and 35 msec, provided that the amplitude of the delayed sound reaching the listener does not exceed the direct sound by more than about 10 dB. If this criterion is met, the directionality of the composite sound will be localized at the primary loudspeaker.

We have seen that for time delays of 50 msec and greater, the listener perceives, under certain conditions, an acoustic gap between the delayed and undelayed signals, which the ear identifies as a discrete, annoying echo. In order to evaluate the extent of echo interference, Haas established certain thresholds of disturbance, the data accumulated in a medium-sized room with an average reverberation time of 0.8 seconds. As shown graphically in FIGURE 3, the percentage of listeners annoyed by the echo (even if the speech material is still intelligible) is plotted against the echo delay time for various echo amplitudes relative to the direct sound. Clearly, an echo of the same intensity as the original sound will cause 20 per cent disturbance for a delay interval of 50 msec. By reducing the echo intensity level 3 dB, the delay time can be as long as 80 msec for 20 per cent disturbance. Most importantly, if

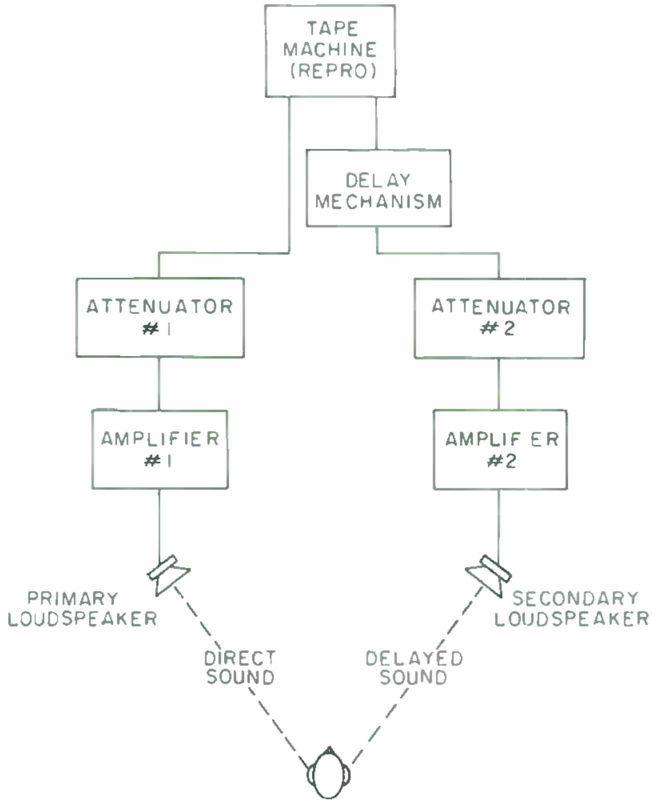


Figure 1. A block diagram of the setup used to demonstrate the precedence effect.

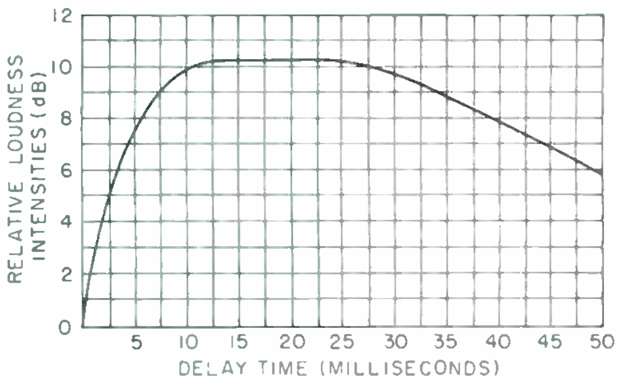
the echo intensity level is more than 10 dB below the direct sound level, it will not disturb the naturalness of speech.

It should be emphasized that these subjective experiments apply to the simulation of a single reflection of a sound source and the results are not strictly comparable with the effect of sound amplification in a hall where a multiplicity of reflections becomes an important factor. Thus, when a secondary echo is delayed beyond the critical period (50 msec), the presence of other echoes in the intervening time diminishes the disturbance of the secondary echo. Under these circumstances, the sequence of reflections useful for intelligibility may extend beyond the 50 msec limit. Nevertheless, the fundamental principles set forth by Haas are still valid, and provide the psychoacoustical basis for the optimum placement of loudspeakers in speech reinforcement systems.

ELECTROACOUSTICAL CONSIDERATIONS

Let us consider how the precedence effect operates in a

Figure 2. The relative intensities of direct sound and delayed sound for equal loudness as a function of delay time (after Haas).



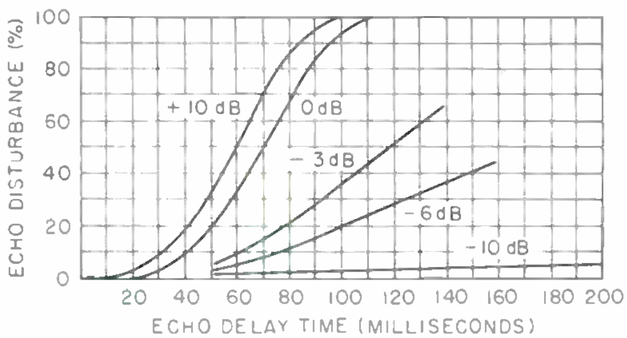


Figure 3. Echo disturbance contours for several echo intensity levels as a function of delay time (after Haas).

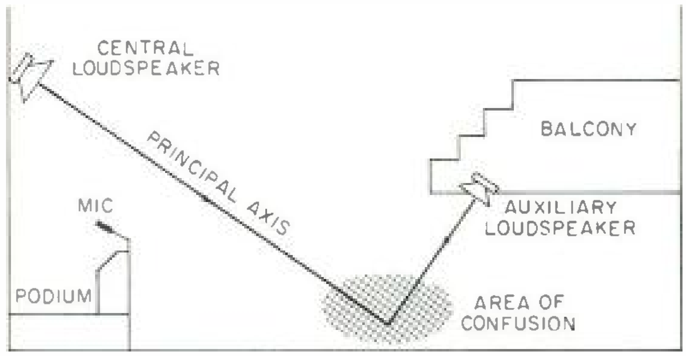


Figure 4. Echo effects produced when reinforced sound arrives at the same area from a different transmission path.

large conference room, where an amplification system reinforces a live sound source bearing a fixed spatial relationship to an audience. Assume that the talker uses a microphone at the center of a podium located at one end of the hall. To preserve directional realism, it would be necessary to arrange the loudspeaker system in such a way that the sound and visual images appear to correspond, i.e., the overall sound apparently coming from the talker with the audience unaware that sound reinforcement is being used.

Suppose, on the one hand, that the sound reinforcement system employs two loudspeakers, one on either side of the hall near the podium area. With this split speaker configuration, many listeners (especially those at the sides) will hear one or the other of the loudspeakers a few milliseconds before they perceive the live sound source. This situation often results in an unnatural aural-visual effect, where the ears tend to concentrate on the sides of the hall, while the eyes focus on the talker at the podium. Consequently, the directional sense of the listeners becomes ambiguous and confused, possibly leading to mental fatigue, over a period of time. In certain cases, where the left-and-right loudspeaker placement is very wide, the sound from the more distant loudspeaker will be heard as an artificial echo of the sound from the nearer loudspeaker, thereby reducing speech intelligibility.

The use of a single-source loudspeaker system, on the other hand, will ensure that sight and sound are similarly oriented. By mounting a centralized loudspeaker (or loudspeaker cluster) directly above the talker, say, 20 or 30 feet, a uniform sound coverage pattern can be provided throughout the audience seating area. High intelligibility and naturalness of speech are achieved because of the similar transmission path length between live sound and reverberant sound, relative to the listeners. In a properly designed system, the directional characteristics of the loudspeaker array are controlled by an amount dependent upon the reverberant characteristics of the hall. According to the precedence effect, if the amplified sound from the central loudspeaker system arrives at the listener's ears slightly after the live sound (between 5 and 35 msec), and the reinforced level is no greater than 10 dB above the direct sound, the live sound will take command and indicate to the listener the direction of the sound source.

The arrival-time effect produced by a single-source loudspeaker system above the talker is aided by the fact that our hearing mechanism is marked by good horizontal resolution but relatively poor vertical resolution; hence localization of a sound source is much more precise in a lateral direction as compared to vertical directivity. Since our ears are located in a horizontal plane, we are thus able to estimate the direction of a horizontally displaced sound source with considerable accuracy. By contrast, vertical displacement is rather difficult to judge with certainty. This means that when a centrally located loudspeaker reproduces acousic energy derived from the live sound source below, the listener instinctively identifies the visible talker as the origin of the blended sound.

TIME-DELAY CORRECTION

In order to attain good intelligibility at the rear of a large hall, the centralized loudspeaker cluster would have to operate at a sufficiently high level without producing dis-

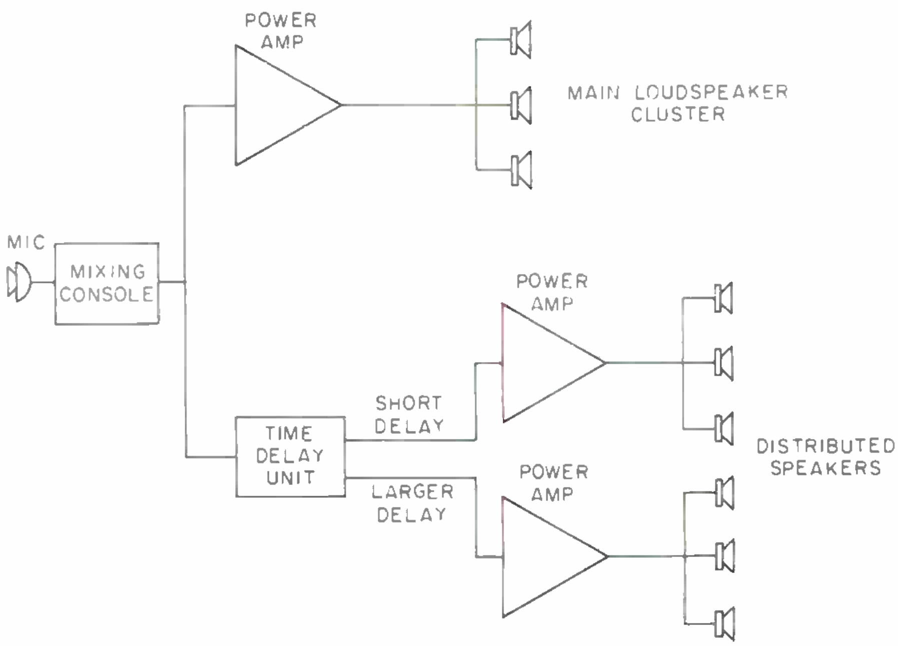


Figure 5. A time delay unit inserted between mixing console and auxiliary power amplifiers.

Figure 6. One of several such products available, this Eventide model 1745A Digital Delay System provides delays up to 398 msec.

turbing echoes from reflecting surfaces. Because of room geometry, it may be necessary to supplement the main system with auxiliary loudspeakers throughout the more distant seating areas, for example, where sound coverage must be provided under deep balconies not reached by the central system. In the example shown in FIGURE 4, the use of a supplementary loudspeaker positioned under a balcony overhang may give rise to an area of interference, or overlap zone, immediately in front of the balcony area. Here the listener will hear first the amplified sound from the nearest auxiliary loudspeaker, followed by the reinforced sound from the main loudspeaker cluster. The resulting interraction between the two sound sources produces a disturbing echo-effect which greatly reduces intelligibility. Furthermore, directional realism is usually lost, especially for those listeners in the back area covered solely by the supplementary loudspeakers.

The time lag problem occurs because acoustic energy in the form of electrical signals travels much faster through cables than sound energy in the form of pressure waves in the air. If, for example, the propagation velocity of airborne sound is taken to be 1130 feet/sec. then the wavefront will advance one foot in 0.885 msec. Thus, for airborne sound to travel, say, 100 feet from the main system to the balcony area, it takes a transit time of 88.5 msec to reach the listener. Assuming that the balcony loudspeaker is 10 feet from the listener, it takes 8.85 msec for this airborne sound to be perceived. Since the propagation time of electrical signals through wires is virtually instantaneous, the listener initially hears the main signal delayed by 88.5 msec, followed by the supplementary signal delayed by 8.85 msec, the time difference being 79.65 msec. Note that the distance between the talker and the microphone is neglected because this transmission path is common to both sound sources.

The precedence effect has established that the interference, or annoyance effect of time-delayed sounds, is influenced by both the difference in arrival times and the relative intensity between the various sounds. Hence, if a suitable time-delay mechanism is inserted into the auxiliary loudspeaker system to produce a delay of 79.65 msec plus, say 20 msec, then the overall sound energy will be additive so as to ensure high intelligibility. Moreover, since the arrival times of both sources will appear to be coincident, no confusion or lack of realism will result. This acoustic illusion will persist even when the sound level of the supplementary signal (time-delay corrected) is greater than the main signal. FIGURE 5 shows a simplified block diagram of a time-delay unit, with two outputs serving distributed loudspeaker groupings at different locations from the main system.

An entirely different approach to loudspeaker placement is the totally distributed system often used in highly reverberent rooms. Instead of a main cluster, a number of loudspeakers are suspended overhead at varying distances from the podium. The loudspeakers are brought within about 12 feet from the listeners' ears, and appropriately spaced to provide even coverage throughout the entire audience seating area. To overcome the time lag problem, time-delay devices are introduced into each loudspeaker system, the delay intervals corresponding to the various distances from the podium. This application of progressive time-delay correction not only improves the apparent acoustics of the hall, but effectively synchronizes the arrival times of the amplified sound with the live sound source. Distributed loudspeaker systems are also utilized in halls where a centralized system is not practical, e.g., low-ceilinged rooms, where a central loudspeaker could not be positioned high enough above the talker to deliver enough gain without the danger of acoustical feedback. In this situation, the loudspeakers are usually flush-mounted in the ceiling, each speaker providing a limited area of coverage to avoid mutual interference.

An alternate scheme employs a comparatively large number of small seat-back loudspeakers operating at low sound levels, and spaced close enough to serve, say, three listeners. Here suitable time-delay units are inserted into the amplifying chain to ensure that all reinforced sound reaching the listeners is perceived simultaneously with the live sound source. Speech signals are so delayed that the live voice from a fixed source passes over the first few rows of seats slightly before the corresponding amplified sound is emitted by loudspeakers in those rows. Similarly, the succeeding rows receive the signals with increased-time delays until the entire room is covered. The time-delay mechanism, in this application, must be capable of producing enough separately controlled delayed outputs for a single input, so that directional realism is assured for all listeners.

TIME DELAY UNITS

In past years, magnetic tape recording systems (employing tape loops or discs) and acoustic-pipe devices have formed the basis for time-delay correction. But owing to the inherent limitation of these methods, the control of time lag in sound reinforcement systems is now being handled by digital delay lines. These recently developed digital systems have been made practical by the application of lsi (large-scale integration) techniques, which allow a multiplicity of complex circuitry to be incorporated into a small space.

In operation, the digital delay line transforms the analog audio input information into digital form, stores it for the desired delay time, and finally re-converts it into an audio output signal for transmission to various loudspeaker systems. A commercially available digital delay line is shown in FIGURE 6. This unit has two independent outputs, each providing up to 199 msec of delays in one msec steps; or, by means of a doubling switch, up to 398 msec in 2 msec steps. In addition, optional modules can be plugged into the unit, not only to increase the number of independent outputs available for any given delay line, but to increase the total delay time available for each output. ■

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MICHAEL RETTINGER

Recording Studio Acoustics

This is the first installment of a six-part series that will appear in these pages in alternate months beginning with this issue.

ALL ENCLOSURES intended for the recording of vocal and instrumental music have certain acoustic elements in common. Recording studios are like people—no two of them alike—and, like people, they can function only under certain common conditions. It is the purpose of the following to discuss some of these requisites.

A recording marred by noise is a recording lost. Rather than exhilerating the listener it may annoy him. Interfering noise is like having a deep scratch on a disk. A sophisticated noise remover can clean up the intervals between the words or the notes but it can do nothing to polish the signals themselves.

It is surprising how few studio designers initiate their work with a noise exposure level measurement of the proposed site, or existing building, if it is to be converted into a studio. Note the words *noise exposure level*, rather than *noise level* measurement. The difference lies in that the former quantity represents a noise history—a variation of noise level with time—while the latter may consist of no more than the observation of a sound-level meter needle deflection during a momentary auditory disturbance at the site.

Every one of the three recent acoustic designs by this investigator of large California recording facilities—the Burbank Studios in Burbank, representing the merger of the Columbia Pictures Corp. and the Warner Bros. Studios; the Audio-Visual Complex of the U.S. Air Force at the Norton Air Force Base in San Bernadino, and the Recording Studios of the U.S. Navy in San Diego—was preceded by a 24-hour graphic level recording of the acoustic climate surrounding the proposed construction area.

Municipal, state, and federal noise exposure level data of a given area are generally in terms of the A-weighted sound level, since this quantity correlates well with people's subjective judgment of the annoyance of many types of noise. This is true also of the *Perceived Noise Level*, which is employed to map airport environment noisiness in terms of PNdBs. What is needed for building insulation requirements is the sound pressure level characteristic and history of the noise, that is, its variation with frequency and its variation with time. One can always obtain the A-weighted sound level characteristic from the sound-pressure level spectrum, but the inverse process is not possible.

Noise histories should be expressed in decile levels ob-

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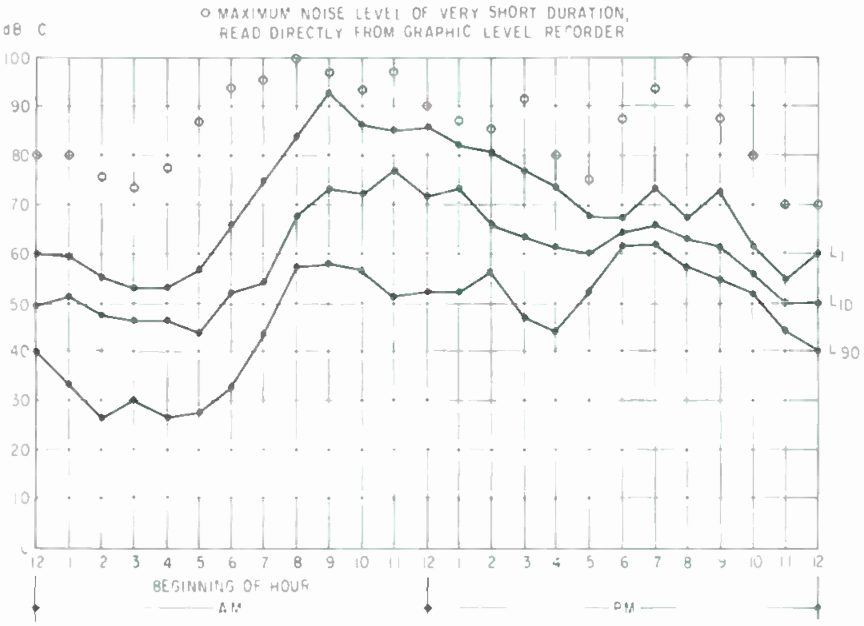

Figure 1. Variation of sound-pressure levels exceeded 1 per cent (L_1), 10 percent (L_{10}) and 90 percent (L_{90}) during the 24-hour noise level history of a proposed sound-recording studio site, together with momentary maximum noise levels, which could possibly be encompassed in a L_1 decile level exceeded 3.6 seconds per hour.

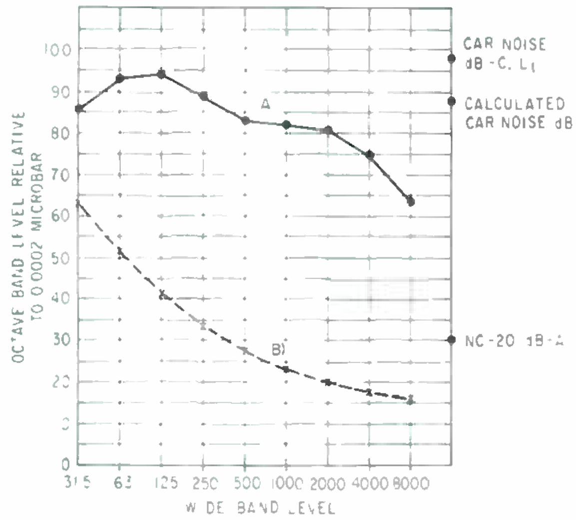

Figure 2. L_1 spectrum of noise at an intended studio locale (A), and Noise Criterion 20, or maximum permissible noise level characteristic in a recording studio (B).

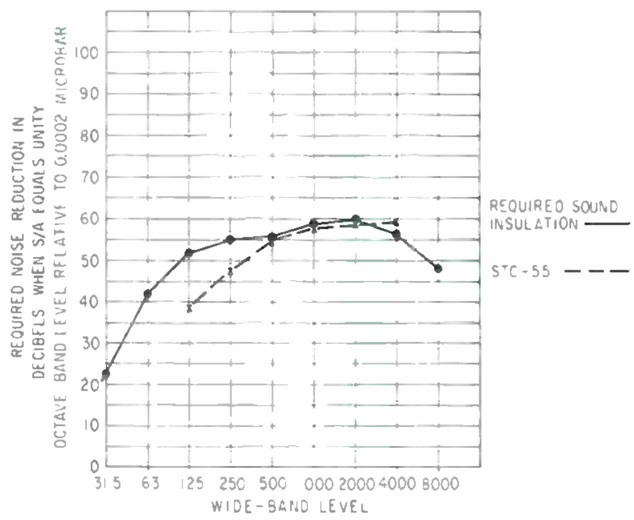

Figure 3. Required sound-insulation characteristic of studio boundaries, graphically determined from the noise level displayed in Figure 2.

tained with a statistical distribution analyzer, and the measurements should be carried out with the "C" weighting network in the common sound level meter. Thus, L_{90} is the decile noise level exceeded 90 percent of the time during the observation period; L_{50}, the level exceeded 50 percent, and L_{10}, that exceeded 10 per cent of the test duration.

FIGURE 1 shows a graph to which both the direct reading and the statistical methods of noise exposure level assessment have been applied in a 24-hour recording at a proposed recording studio site. In the evaluation of such information one must note, for instance, whether a given L_1 level transmission into the studio can be accepted 1 percent of the time, that is, 36 seconds per hour, or whether an acoustic compromise cannot be allowed. While such an intrusion may not be permitted in a recording studio, it may but rarely be responsible for a bad "take" in a motion-picture sound stage where scenes are photographed once every two hours or at even greater intervals. For this reason, there are still film-recording studios in Hollywood which permit three-times daily low overflights of traffic-reporting helicopters, which can be heard inside the building.

Curve (A) of FIGURE 2 shows the L_1 spectrum of a noise environment whose wide-band sound-pressure level is 98 dB-C and whose calculated A-weighted sound level comes to 88 dB-A, which conforms to the California Vehicle Code noise level limit of a 35 mile/hr. truck at 50 feet.

Curve (B) of FIGURE 2 represents the NC-20 criterion, or maximum acceptable noise level characteristic inside a recording studio, whose wide-band sound level is 30 dB-A.

The difference between curves (A) and (B), shown on FIGURE 3 constitutes the required sound insulation characteristic of the studio's external boundaries under the restriction that S/A, the ratio of the boundary surface to sound-absorption in the studio, is unity (a frequently-met condition in such structures) and that all boundaries are exposed to the same noise level.

Because the 500 hertz noise-reduction necessary for the walls and roof of such a building comes to 55 dB, the less knowledgeable architects and designers are tempted to specify a sound insulation rating of STC-55 (Sound-Transmission Class 55) for the boundaries. However, as drawn in on FIGURE 3 with a dotted line, such a sound attenuation would be inadequate for these space-dividers. Of course, the ASTM (American Society for Testing and Materials) notes in its standard E413-70T) that excluded from the scope of this classification system are the exterior walls of buildings, whose noise problems are most likely to involve motor vehicles or aircraft.

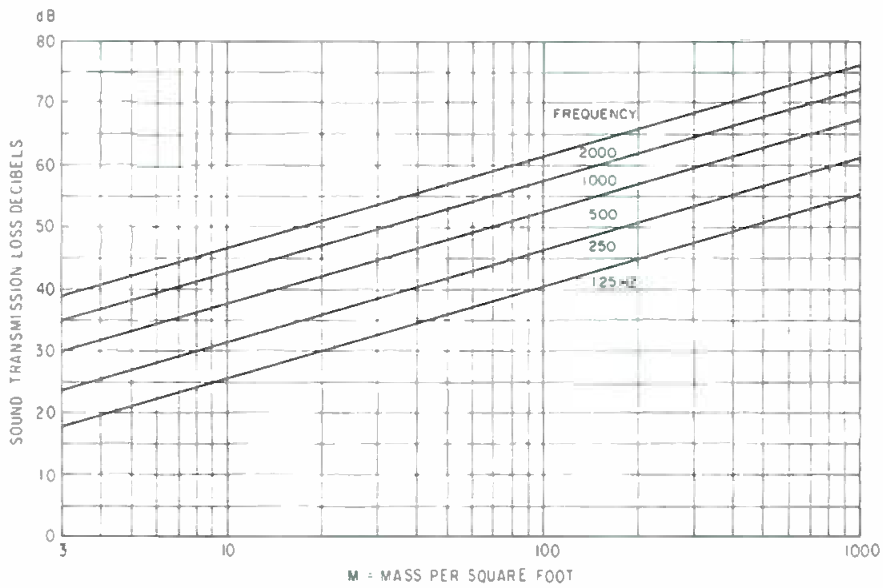

Figure 4. Acoustic Mass Law, or variation of sound-transmission loss with surface density (mass per square foot) of building boundary, for several frequencies.

FIGURE 4 illustrates the so-called *acoustic mass law*, an empiric finding, proposing that the mid-range, or 500-hertz sound-transmission loss, of a barrier varies according to

$$TL = 23 + 14.5 \log M$$

where M = surface density of wall, lbs./sq. ft.

Accordingly, a monolithic barrier, to exhibit a sound-transmission loss of 55 dB at 500 hertz (as called for by the curve shown on FIGURE 3), would have to have a surface density of 160 lbs./sq. ft. Since this represents a concrete wall more than 12 inches thick, recourse is taken to double barriers. A discussion of the sound-insulation of double walls will be found in the author's book *Acoustic Design and Noise Control*, available through the office of **db**.

A low noise level in a recording studio is also dependent on the type of air-conditioning system equipment installed in the building. For such rooms, an air-velocity of no more than 500 ft./min. is recommended, because higher air speeds invariably produce turbulency at the diffusers. This condition cannot be corrected after construction by the introduction of duct silencers, or short duct sections with sound traps. The reason is that the noise is not a fan hum carried through the line, but aerodynamic hiss at the duct ends. The size of a required duct depends on the number of air-changes per hour desired in the studio. A high number of such changes in a large studio, like 10 changes per hour in a room of more than 50,000 cubic feet volume, calls for relatively large ducts when the air velocity is low (for which reason subsequent alterations are frequently impossible because of space limitations in the walls).

A studio also must look right to be right. I well remember the time I was called to a major motion picture studio in which the diva had refused to sing because the stage looked, she had said, like a cow barn. The supervisor of the sound department had quickly converted a sound stage (a large stage in which sets are erected and photography takes place) into a music- and song-recording locale by placing a number of fir plywood panels about the walls and near the ceiling. According to his measurements, the acoustics of the room were excellent, and some sample recordings made by his staff prior to the recording session substantiated his results. Yet, the musicians were not happy and the diva was furious. It took me, a well-known interior decorator, and a working crew of six, the better part of two days to convert the locale to an esthetically pleasing and acoustically satisfactory recording milieu.

I was also called into a new recording studio which featured hand-rubbed hardwood panels, ¾-inch plush carpet, and other elegant trappings which made a beautiful interior decor but in which the recordings left a great deal to be desired even afer hours of experimental microphone placements for best sound pick-up. The brave but inexperienced designer of the studio had oriented the stage panels incorrectly, so that instead of providing a suitable efflux for the music, the sound was sent back into the stage to mill around therein for the confusion of the musicians and to the detriment of the recordings. Surprisingly, by merely positioning the band at the opposite end of the stage, the recordings were much improved. Finally, the entire room had to be re-worked, at no little expense. The moral of the story is that it is much cheaper to do it right in the first place.

Other common studio requirements will be discussed in the next installment. ■

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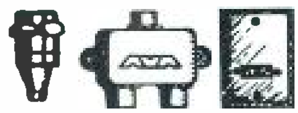

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PEOPLE, PLACES, HAPPENINGS

db HAS MOVED

Sagamore Publishing Co. Inc. has finally had to move to larger quarters. We held out for as long as we could, but when the publisher walked in one day and found his desk appropriated by a new secretary he got the message.

Our new "luxurious" offices are only a short distance away at:

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We have been able to retain the same telephone number—**(516) 433-6530.**

If you are in the area we would be delighted to have you visit us.

● Three executives have been promoted at **Shure Brothers, Inc. of** Evanston, Illinois. **Raymond E. Ward** assumes new duties as vice president, sales and marketing promotion. **Howard T. Harwood** has been named manager, marketing promotion, in charge of all advertising and public relations. **Norman A. Hesslink, Jr.** has been promoted to advertising manager, a liaison post between the company's own advertising department and its advertising agency.

● Expansion of its sales operations has been marked by the **Scully/Metrotech** recording divisions of **Dictaphone Corporation,** Mountain View, California, with the establishment of a new position, national sales manager. **Homer Hull** has been appointed to the post. Mr. Hull has been with Scully/Metrotech since 1972. Prior to that he was with **Ampex Corporation.**

● A novel sales presentation idea, called the **Suitcase Seminar** and originated by **James B. Lansing Sound, Inc.,** has branched out from its original intent and found interest in educational applications. The presentation consists of cutaway speaker components, flip chart, and audio-visual equipment, all packed into a standard size aluminum suitcase. Recently, **Tom Frisina** and **Ron Cotterell** were invited to the **University of California at Santa Barbara's** *Introduction to Audio* class. Their vivid presentation, with the aid of the suitcase demonstration, was well received for its effectiveness.

● As we went to press with this pre-AES show issue we have been saddened to learn of the following deaths:

Dorothy Spronck, administrative secretary of the **Audio Engineering Society,** succumbed to the effects of a stroke which she had suffered earlier. Death occurred on July 29.

Howard Holzer, president of **Holzer Audio Eng. Co.,** was killed in a Mexico City airplane crash. Mr. Holzer, who was an experienced pilot, was at the controls of a private plane at the time of the crash.

William (Bill) Hazlett), who many remember as **Altec's** New York office man, died in late July. He was born in 1901. He was a life charter member of the AES since 1970.

We knew and worked with these three people. We feel their loss personally and know that this emotion is shared by many.

● **Thomas Creighton III** has been appointed director of sales and marketing by **Broadcast Electronics, Inc.** of Silver Springs, Maryland. Mr. Creighton will supervise the sales of the Spotmaster line of tape cartridge machines, audio consoles, and related audio products. Mr. Creighton was formerly associated with **McMartin Industries.**

● According to a new **FCC** ruling, f.m. stations will be permitted, without notification or application to the Commission, to incorporate a combination of Dolby B-Type noise reduction and reduced pre-emphasis (25-microseconds instead of 75-microseconds) in their transmission. The system, devised by **Dr. Ray Dolby,** reduces pre-emphasis, the amount by which high frequencies are boosted during transmission without the handicap of dullness. The system is entirely compatible to existing equipment. It can be received with some improvement in quality by consumer receivers not equipped with the noise reduction units and with greatly heightened effectiveness by those consumers whose sets are equipped with the Dolby system.

● A new firm, **Joel Associates,** of Teaneck, N.J., has been formed by **Irving Joel,** formerly chief engineer with **A & R Recording, Inc.** Mr. Joel will maintain his connection with A & R as a special projects consultant. He has been responsible for the design and construction of numerous recording and sound reinforcement systems and special equipment for theater and night club entertainers, as well as maintenance for recording studios. Joel Associates will introduce a new audio test equipment line in the fall.

If you work with microphones, you need this book!

The most important microphone book ever published.

COVERS EVERY SIGNIFICANT ASPECT OF THEORY AND USE FROM A TO Z!

At last, the whole field of microphone design and application has been prepared and explained in one concise, fact-filled volume by one of audio's outstanding experts. This book is complete, up-to-the minute and so full of useful information, we think you'll use it every time you face a new or unusual microphone problem.

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The twenty-six fact-packed chapters in this indispensable volume cover the whole field of microphones from theory, physical limitations, electro-acoustic limitations, maintenance and evaluation to applications, accessories and associated equipment. Each section is crammed with experience-tested detailed information. Whatever your audio specialty – you need this book!

Along with down-to-earth advice on trouble-free microphone applications, author Lou Burroughs passes on dozens of invaluable secrets learned through his many years of experience. He solves the practical problems you meet in day-to-day situations. For example:

* **When would you choose a cardioid, omni-directional, or bi-directional mic?**
* **How are omni-directional mics used for orchestral pickup?**
* **How does dirt in the microphone rob you of response?**
* **How do you space your microphones to bring out the best in each performer?**

This text is highly recommended as a teaching tool and reference for all those in the audio industry. *Price: $20.00*

THE AUTHOR

Holder of twenty-three patents on electro-acoustic products, Lou Burroughs has been responsible for extensive contributions in the development of the microphone. During World War II, he developed the first noise cancelling (differential) microphone, known as the model T-45. Used by the Army Signal Corps, this achievement was cited by the Secretary of War. Burroughs was the creator of *acoustalloy*, a non-metallic sheet from which dynamic diaphragms are molded. This material made it possible to produce the first wide-range uniform-response dynamic microphone. Burroughs participated in the design and development of a number of the microphones which have made modern broadcasting possible – the first one-inch diameter wide-range dynamic for tv use; the first lavalier; the first cardiline microphone (which ultimately won a Motion Picture Academy award) and the first variable-D dynamic cardioid microphone. He also developed the first wind screens to use polyester foam. Burroughs was one of the two original founders of Electro-Voice, Inc. He is a charter member of the Society of Broadcast Engineers and a Fellow member of the Audio Engineering Society.

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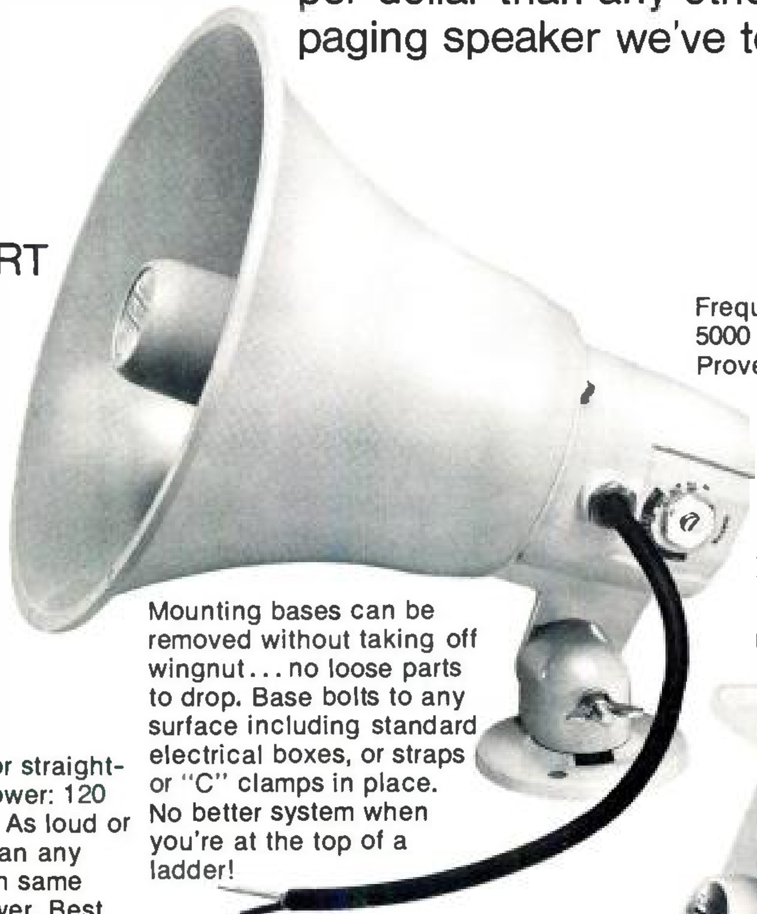

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